UNDERWATER COMMUNICATIONS SYSTEM
FOR USE BY FREE SWIMMING DIVERS

David Campbell Steere

NAVAL POSTGRADUATE SCHOOL
Monterey, California

THESIS

UNDERWATER COMMUNICATIONS SYSTEM
FOR USE BY FREE SWIMMING DIVERS

by

David Campbell Steere

June 1975

Thesis Advisor: George H. Marmont

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The need for a compact, reliable, and inexpensive communications system for untethered SCUBA divers is pointed out as being acute. An investigation into various characteristics of the undersea medium leads to a choice of a 40 kHz acoustic carrier, frequency modulated by plain voice, as the optimal approach. Investigations into microphone, earphone, and physiological problems encountered in the diver's environment

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Underwater Communications System For Use By Free Swimming Divers

by

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Lieutenant, United States Navy
B.S., United States Naval Academy, 1968

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requirements for the degree of

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TABLE OF CONTENTS

LIST OF TABLES

LIST OF ILLUSTRATIONS

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I. NEED FOR UNDERWATER COMMUNICATIONS

A. BRIEF HISTORY

Since man's first attempts to swim underwater, a very strong need has existed for him to communicate--not only with his fellow swimmers but also with his compatriots ashore or on the surface. Unlike the wide variety of applications of diving in today's society, most ancient diving exploits were connected with naval warfare. For example, Xerxes is said to have used combat divers, and over 400 years before Christ, Herodotus recorded into written history accounts of Scyllis, the famous Greek diver who not only recovered vast amounts of treasure from sunken Persian ships, but also cut the anchor lines of an entire fleet during a violent storm, then swam nine miles to Artemisium and safety. Although the early records make little mention of attempts to communicate with divers such as Scyllis, there is little doubt that a useful communication system with his commander would have made his work safer, more efficient, and the outcome more certain.[1]

Little has been recorded of a scientific approach to utilizing the underwater medium for the tactical information that it carried or for communication purposes until 1490 when Leonard di Vinci stated: "If you cause your ship to stop, and place the head of a long tube in the water, and place the

[1]NAVSHIPS 0994-001-9010, U.S. Navy Diving Manual, p. 7-10, March 1970.

outer extremity to your ear, you will hear ships at a great distance from you." It was not until 1827 that the Swiss physicist Daniel Colladon made the first quantitative underwater sound measurements. Finally, in 1880, transduction--the process of conversion of electrical energy to sound waves, was developed by Jacques and Pierre Curie in their work with piezoelectric crystals.[2]

Since that time great and rapid strides have been made in electronics and communications engineering. Concurrently, due mainly to urgent wartime requirements and the development of SONAR, much has been learned about the ocean environment and its characteristics as a transmission medium. However, relatively little has been accomplished in both these fields simultaneously. That is, recent design breakthroughs in communications have allowed the development of flawless voice and video transmissions as well as data links from points in outer space. At the same time, modern underwater acoustics engineers have designed and deployed capable and advanced detection and localization SONAR schemes.

B. OVERVIEW OF SYSTEM REQUIREMENTS

The need for a good communication system by today's SCUBA diver is unquestionable. This need might vary from a matter of convenience for a pair of spear fishermen, or of expedience for a Navy salvage crew, to the very means of survival in a clandestine military reconnaissance operation. It appears

[2] Urick, Robert J., Principles of Underwater Sound, p. 2-3 McGraw Hill, 1967.

that the most urgent need for such a system is being experienced by today's U.S. Navy divers. However, the proposed system has obvious applications in commercial and sport diving communications as well.

Discounting purely research efforts such as the AQUANAUT program, U.S. Navy diving operations today consist of three major components: Salvage, Explosive Ordnance Disposal, and covert warfare teams such as SEAL and UDT. All three of these groups of divers make extensive use of un-tethered diving equipment, most predominently of the class of diving operations known as SCUBA (Self-Contained-Underwater-Breathing-Apparatus). This equipment is inexpensive, easy to store and maintain, and provides a degree of flexibility that is not likely to be made obsolete by design breakthroughs in the very near future. It is, however, doubtless that countless salvage efforts have been unsuccessful, or at the very least, time-consuming, because of the lack of ability for SCUBA divers to communicate even the simplest of thoughts with other divers or the surface support crew. Obviously, the same can be said of EOD and UDT/SEAL operations.

Outside of military requirements, a reliable communications system for SCUBA diver use would doubtlessly increase the safety and enjoyment of thousands of sport divers. Furthermore, salvage and underwater construction workers involved in a myriad of various diving activities would also be very enthusiastic about the adoption of such a system if it could be developed.

The communication system must be capable of providing a means of accurately expressing simple thoughts such as "I am OK," or, "Do you need assistance?" In addition, it must allow for the rapid expression of relatively complex exchanges of thoughts such as "Lower to me by cable the following tools: ...." Although for many types of diving operations a simple continuous wave (CW) pulse scheme might represent a significant improvement over hand signals and tugs on a tending line, this system would severely limit the expression of any but expected exchanges of thought. For this reason as well as the multipath problem addressed below, a simple CW scheme has been discarded as a valid approach to the problem.

Recognizing the difficulty of installing a microphone in a wet environment characterized by very high ambient pressure, a second approach was investigated. Standard electrode lead-offs can be attached to some area of the diver's body that is relatively devoid of muscle activity, such as behind the ears. Through a process of bio-feedback, the diver could learn to send electrical signals to the vicinity of the electrode in exactly the same way he sends muscle commands by nerve conduction of action potentials to his arms and legs and other muscular areas. This form of voluntary stimulation could be converted to electrical currents at audio frequencies and then be further processed and used to modulate a carrier wave which is transmitted through the ocean medium. Such an arrangement was attempted in the laboratory and although numerous distinctive sounds could be made simply by willing them to occur,

repeatability of the sounds takes a considerable time to learn. Furthermore, the use of electrodes in the salt water environment presents severe difficulties.[3] It was for these reasons and several others that this approach was abandoned.

It will be necessary, then, to provide a reasonably faithful replica of the sender's voice message to the other divers as well as the surface personnel associated with the operation. This message must be suitable for transmission over distances less than one meter to fellow divers as well as the entire distance to the support personnel on the surface. An absolute physiological limit for operating depth of a SCUBA diver is 100 meters. For safety purposes, U.S. Navy SCUBA operations are constrained to a maximum depth of 130 feet (39.6 meters). Due to human endurance and exposure limitations as well as the frequent need for direct support from the surface, latteral ranges in excess of 500 meters from the surface unit are not common. Slant range in excess of 510 meters would be rarely exceeded. Thus a one kilometer maximum range is seen as more than adequate for virtually all Navy operations and is therefore used as a design criterion.

As has been discussed above, the system required by today's Navy divers should be voice-modulated and capable of a one thousand meter range. From a practical viewpoint, the system

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[3] Hamilton, John D. M., A Diver Monitor System, U.S. Naval Postgraduate School, Monterey, California, 1974.

must also be acceptable to the user. As such, it should be characterized by the following:[4, 5]

1. Easy to operate, even when wearing bulky gloves.
2. Readily maintained with off-the-shelf spare parts and relatively simple technology.
3. Reliable and durable, and inexpensive.
4. Safe to the user. The probability of system-induced detonation of explosives must be low.
5. Compact. The system must be engineered for minimum encumbrance to the diver.
6. Battery life of at least two hours. Rechargeable batteries.
7. Wide AGC range.
8. Squelch circuitry.
9. Means of monitoring own transmitted signal.

It is felt that the system described below meets these objectives.

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[4]Beagles, John A., <u>Some Basic Problems in Diver Communications</u>, p. 1-6, Naval Undersea Center, San Diego, 1970.

[5]Miller, C. N., <u>Needed Development for Swimmer Communications</u>, p. 1-5, Naval Undersea Center, San Diego, 1969.

II. PROBLEMS ASSOCIATED WITH THE DESIGN OF A DIVER COMMUNICATION SYSTEM

The very nature of the diving environment presents several unique problems to the designer of communications equipment. In fact, the problem of communicating between SCUBA divers is an interdisciplinary one. Some knowledge of human physiology, acoustics, mechanical and electrical engineering, and some familiarity with SCUBA diving techniques are all necessary for the overall design. Seldom has the communications engineer been faced with such a range of seemingly difficult problems simultaneously. Some of these problem areas are discussed below:

A. PHYSICAL ENVIRONMENT

Two of the most obvious problems inherent in the diver's environment are lack of light and warmth. Frequently, water temperatures dictate the use of bulky and cumbersome equipment including hood, gloves, and full wet suit. The Navy diver usually works in total darkness and must identify controls, tools, and other objects strictly by feel. The ideal communication system for his use would require no tuning and simple transmit-receive switching.

Within the Navy-approved depths, the enclosure for the communication equipment and all parts external to the enclosure must be capable of withstanding ambient pressures up to five atmospheres. Other SCUBA divers may on occasion operate at

even greater depths and correspondingly higher pressures must be within the capability of the equipment.

Temperature range, although not excessive, must be considered. It is probable that the vast majority of SCUBA operations will be conducted within the range 0-40°C.

Finally, like all other SCUBA-related equipment, the communications gear will undoubtedly be subjected to moderately severe shock. Aside from careless topside handling, it is likely that the diver, limited in his ability to see, will come in contact with some solid underwater object and the equipment must not be so fragile as to be easily damaged by such accidents.

Widespread use is made of two types of air demand regulators under the general category of SCUBA equipment. These are the single hose and double hose. The double hose regulator has been in existence longer but is generally considered outmoded by the newer single hose type. For the purposes of this discussion, the major difference between the two types of regulators is that with the single hose regulator, exhaust air is released as bubbles only about three centimeters from the diver's mouth at about chin level. For most double hose regulators, it is probable that slightly less back-pressure is exerted on exhaust and more significantly, bubbles are exhausted behind the head, well removed from the diver's mouth. It is thus believed that the double hose regulator would probably prove superior for the purposes of communication if other than a full-face mask is utilized.

The problem common to both standard regulators is two-fold. First, a portion of each type extends into the mouth and is held in place by the front teeth. Thus, the lips cannot be fully closed and many consonant sounds which utilize the teeth, such as "th" and "s," are impossible to make normally. Furthermore, nasal sounds are eliminated altogether by placing the speech cavity before the mouth only. More importantly, though, both the single and the double-hose regulator present a severe speech distortion because of the necessarily small chamber placed in front of the lips. This, coupled with the tremendous impedance mismatch of most of the rubber materials used in the speech chamber, causes the speaker's intelligibility to be badly distorted before any electronic processing is started.

The double hose "muzzle" with built-in microphone eliminates the need for biting by using a neck strap to hold it in place. Unfortunately, the speech chamber in front of the mouth is still quite small and unnatural resonances are likely. The Nautilus (trademark) muzzle made by the Bioengionics Corporation with a Hydro Products (Aquasonics) model MS microphone falls under this category and is used in testing stages of the proposed system.

The full face mask provides a much larger chamber into which the diver can articulate his words. Although air noise (supply and exhaust) could be a real problem with the full face mask, this scheme is seen as an improvement over the muzzle scheme mentioned above. The SCUBA-PRO model Visionaire full face mask with a Navy Coastal Systems Lab microphone is utilized in the evaluation of the proposed system.

B. MAN-TRANSDUCER INTERFACE

1. Microphone

The acoustic energy associated with normal speech is surprisingly small; during one second at conversational levels about 200 ergs of energy are produced. Speech intensities produced at normal conversational levels generally have a dynamic range of greater than 700:1. The vowels are characterized by strongest intensities but even these have roughly a three to one range of intensity. For example, the intensity of the sound "aw" as in "talk" is three times that of the weakest vowel sound, "ee," as in "see."

Little is known of the effects of increased ambient pressure and back-pressure provided by exhaust valves on the breathing apparatus upon the dynamic range of speech of the underwater communicator. Nevertheless, the range is probably somewhat less than that of the speaker in a large room filled with air at atmospheric pressure.

Figure 1 depicts a long-time average spectrum of normal speech. It shows that speech energy is generally restricted to frequencies between 50 and 7,000 Hz. The energy is greatest in the 100-600 Hz region, the range of fundamental frequencies of a typical male speaker.[6]

Although the author has been unable to locate any quantitative justification, it is generally agreed that the

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[6]Denes, Peter B., and Pinson, Elliot N., The Speech Chain, p. 116, Bell Telephone Laboratories, March 1969.

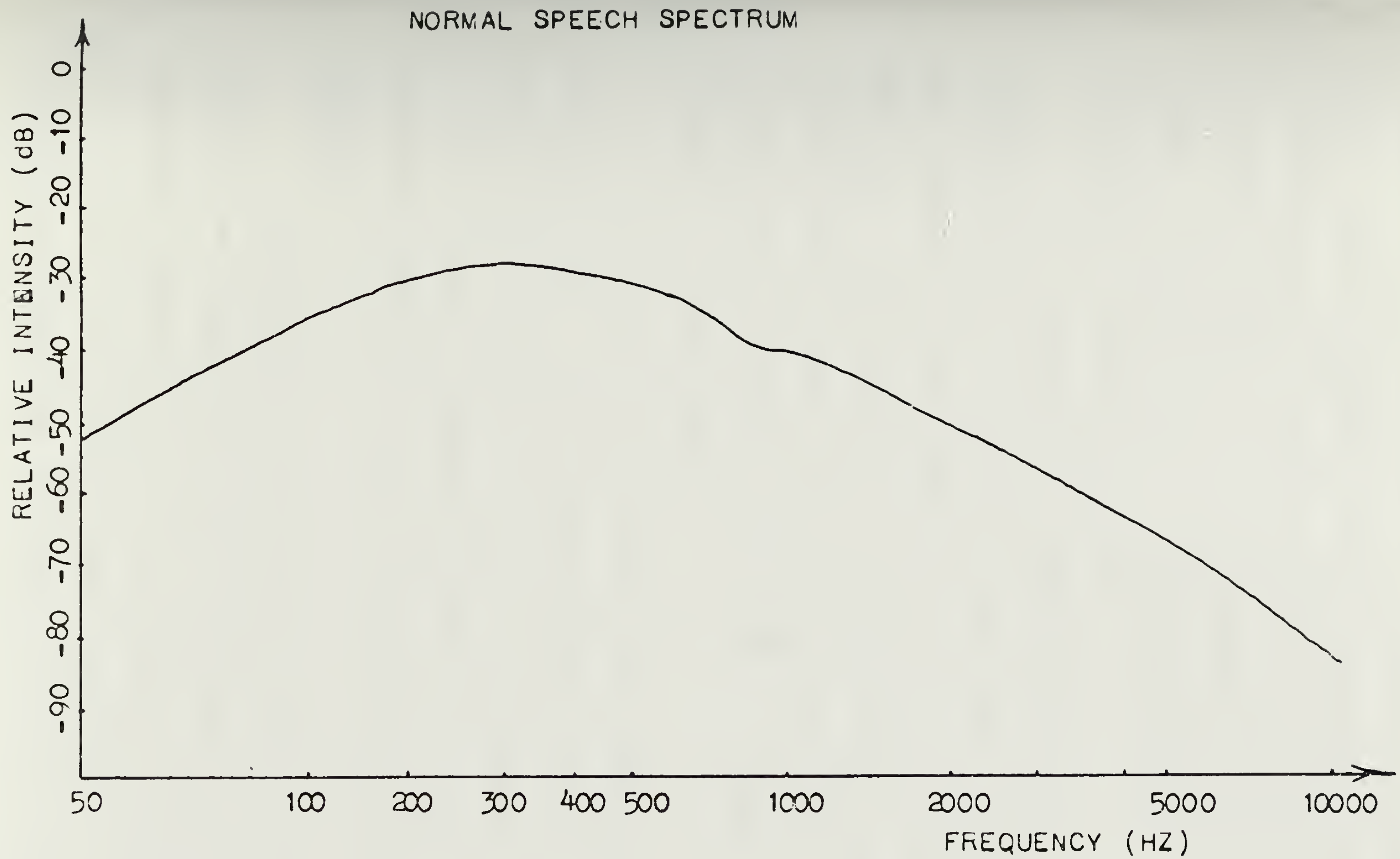
NORMAL SPEECH SPECTRUM
RELATIVE INTENSITY (dB)
0
-10
-20
-30
-40
-50
-60
-70
-80
-90
50
100
200
300
400
500
1000
2000
5000
10000
FREQUENCY (HZ)

FIGURE 1

human speech-making apparatus, when subjected to increasing pressure, shifts its output frequency upward. [2]. Hence, adherence to the generally accepted telephone standards (3 kHz = upper frequency cutoff) would be a serious mistake for the designer of a viable underwater communication system. It is the concensus of several experts in the employ of the U.S. Navy [2] that an upper frequency of about 7 kHz would be a far better design criterion.

The mechanism of human speech can be modeled by an audio signal generator with some means of projecting its output energy as sound pressure waves, and a series of resonant filters. These parts of the model would correspond to the vocal chords, and a series of cavities (of various characteristics and resonances) respectively. Figure 2 provides a rough depiction of this model.

The series of cavities which make up the filter depicted by h(t) is a complex modifier of the sound projected by the vocal chords. By a very complicated and not easily learned process of muscle control, the shape, size and backpressure provided by these cavities can be modulated to provide speech.[7]

By placing still another cavity in series with the diver's own vocal cavities, the filter transfer function of the human speech apparatus is modified. Since it is necessary for the diver to articulate his words into a separate cavity with its own inherent resonances and back pressure, the

[7] ibid., p. 39-65.

HUMAN SPEECH MODEL

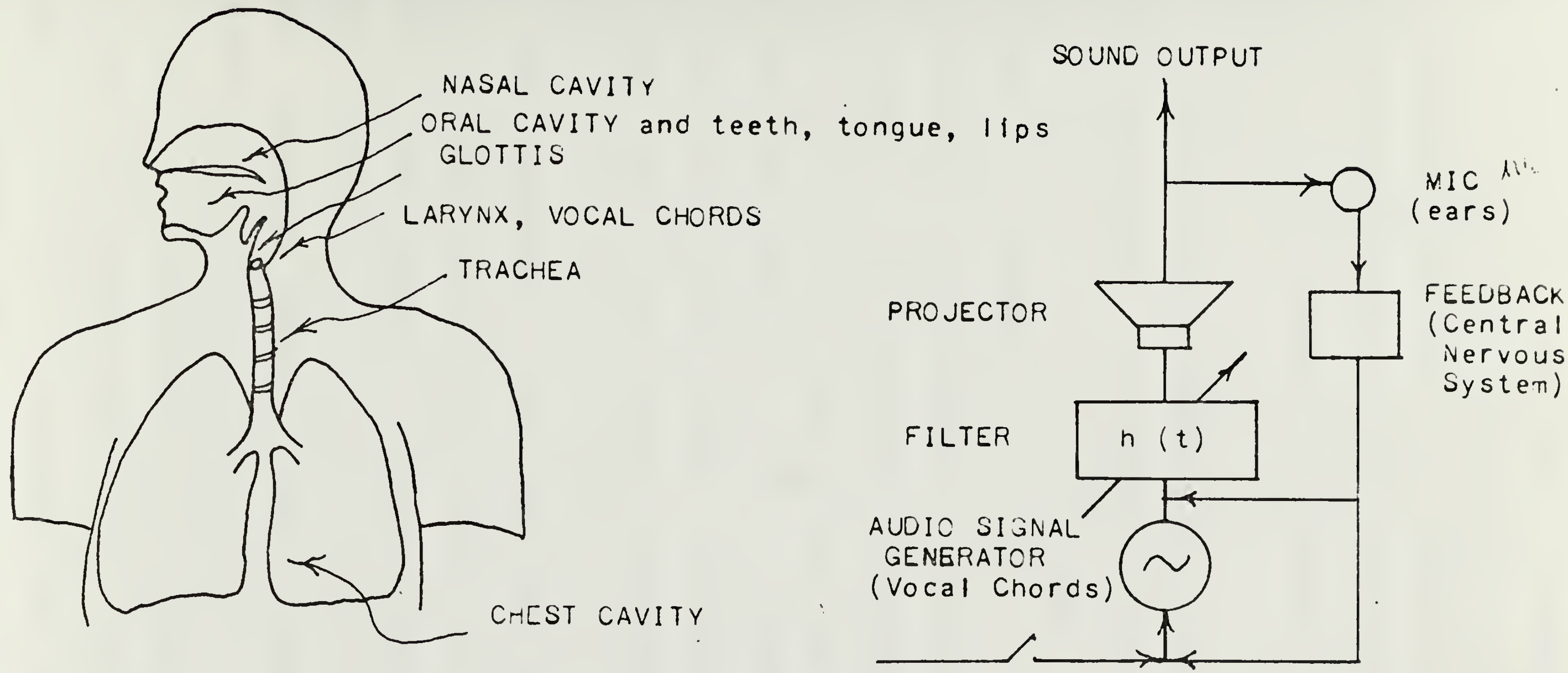

FIGURE 2

characteristics of his speech may be altered. It is possible, however, that if by some means the diver were able to monitor his own outgoing signal that, by a process of bio-feedback, he might be able to learn to reduce the distortion caused by this effect. This technique is utilized in the design of the proposed system.

A computer-aided design of this cavity would be a very worthwhile undertaking. The end goal would be the minimization of the distortion by optimizing the shape of the speech cavity and the material from which it is made. It is doubtful that such a design has been carried out by other than a trial-and-error technique. It is suspected that the use of a full face mask providing maximum possible volume into which the diver could speak would prove to be the best approach to the primitive selection of a cavity shape. Another approach is the muzzle which suffers the inherent disadvantage of separating the nasal sounds from the microphone cavity.

The microphone has been named the weakest link in the underwater communications chain. Aside from the necessary expansion of high frequency response characteristics to accommodate the change in the diver's speech spectrum, and the unnatural resonances imposed by the microphone cavity, numerous other problems plague microphone design.

Except for a few specialized microphone applications in very high noise areas such as flight deck communications, it is hard to conceive of a more difficult environment in which to obtain good signal-to-noise ratios out of a microphone.

Bubble noise could probably be accurately modeled as Gaussian White Noise over the speech spectrum with high mean voltage.[8] See Figure 3.

Probably the most severe problem associated with virtually all forms of microphones is their susceptibility to changes in characteristics with increasing pressure. Assuming a microphone is made totally impervious to moisture (occasional total immersion in sea water would certainly be expected of any SCUBA-related microphone), high ambient pressure may influence the properties of many types of microphone.

Applied Research Laboratories, Naval Electronics Laboratory Center, Naval Undersea Center, among others, have made extensive tests of various microphones in the diving environment.[8] Throat microphones have scored low in intelligibility tests because of almost total loss of lip and tongue-formed consonants. Bone conduction microphones have proven to be susceptible to bubble noise. Although further testing is probably indicated, in general, lip microphones have performed best under a wide variety of applications.

Recent developments in the diaphragm-type gradient microphone have been very promising. Dr. Charles T. Morrow of Ling Tempo Vought has developed a number of microphones of this type that are pressure insensitive, directional, and noise cancelling. Unfortunately, these microphones are not in mass production, and their cost, at present, makes them unsuitable for a low cost system.

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[8]Thompson, Lewis A., Underwater Wireless Communication, p. 24, Applied Research Laboratory, University of Texas at Austin, January 1974.

BUBBLE NOISE SPECTRUM

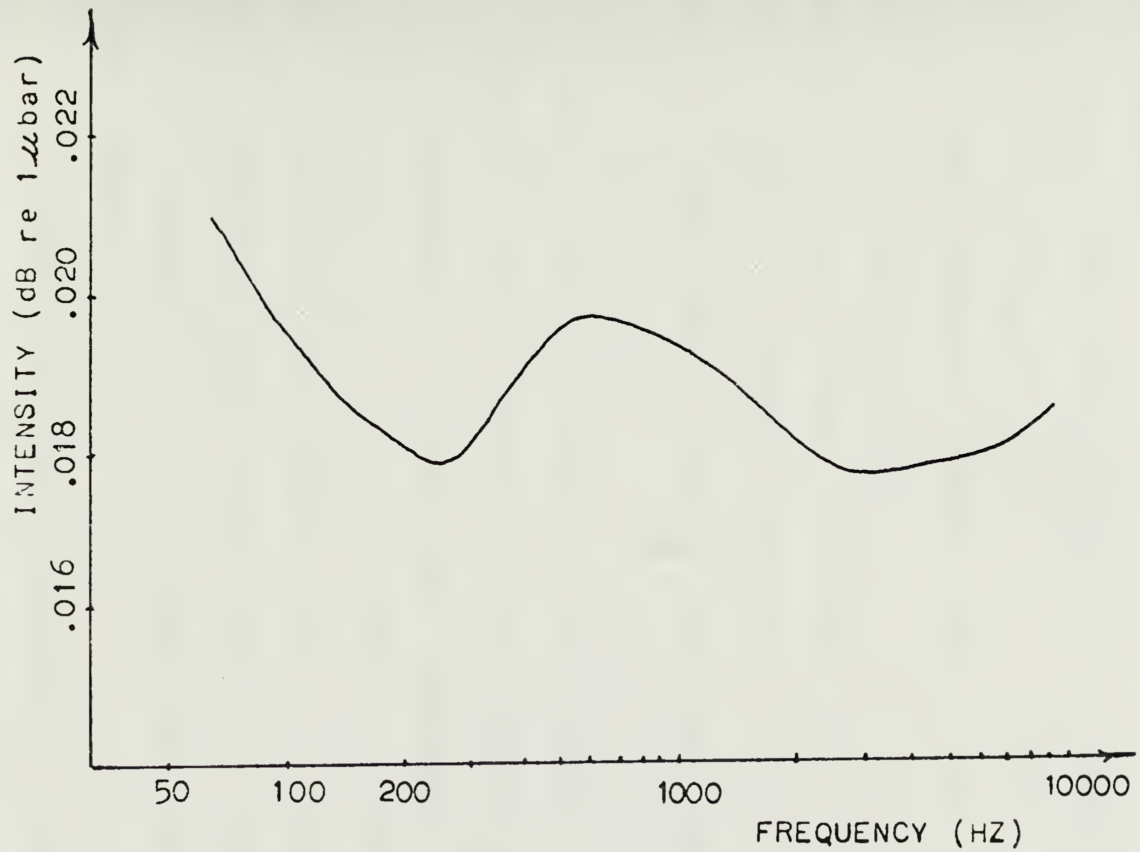

FIGURE 3

A second microphone of excellent performance has been a ceramic-disc transducer developed by the Navy Coastal Systems Laboratory, Panama City, Florida. This microphone is one of two which are utilized in the communications system described here.

2. Earphones

When the human hearing apparatus is immersed in water, severe loss of hearing threshold is experienced. This is because of a large impedance mismatch which occurs. By allowing passage of water into the meatus, and into contact with the external side of the eardrum, the middle ear, filled with air, provides an entirely different impedance to the inside of the eardrum. The three bones (maleus, incus, and stapes) in the middle ear normally set up a 1.3 : 1 mechanical advantage to conduct sound to the oval window of the cochlea. This mechanical arrangement which performs very well when the meatus is air-filled, provides very ineffective coupling of sound pressure waves to the cochlea when the meatus is filled with water. Several investigators have suggested that the entire outer and middle ear mechanisms are, for the most part, bypassed and that the major process by which hearing takes place underwater is by bone conduction.

It is necessary for water to enter the meatus to enable the diver to equalize pressure on both sides of the eardrum. Most experienced divers would be very reluctant to place any type of object in their ears.

It is for these reasons that bone conduction earphones represent the best approach to presenting the demodulated audio

to the diver. The bone conduction earphone used in the AQUA-SONIC model U420 communicator is used in this system.

III. THE UNDERWATER ENVIRONMENT AND SELECTION OF A MODULATION MODE

Since the early scientists first began their study of the underwater world, several physical characteristics of the environment have consistently shown themselves to be problematic. First, sea water is a relatively dense medium and, as such, provides severely high pressures at moderate depth (approximately .445 pounds per square inch per foot of depth). Additionally, sea water can be quite corrosive to many construction materials. Attenuation to acoustic waves is severe in the undersea environment. Furthermore, the highly conductive nature of the ocean causes it to be rather unsuitable as an electromagnetic transmission medium. Also, a primary consideration in this study is that of noise; the underwater environment is one of the most difficult media known for transmission of information because of its exceptionally high noise level.

In order to convey useful information from one point in the undersea environment to another, energy of some form must be modulated by the information content and transmitted to a suitable receiver. The two most commonly utilized forms of this energy are acoustic (longitudinal waves in the audible and ultrasonic portions of the spectrum) and electromagnetic (radio waves or light energy).

For diver communications, electromagnetic radio propagation is rendered almost totally infeasible. A thorough

discussion of the use of electromagnetic radiation underwater[9] indicates that under ideal conditions with a ten watt transmitter and an "optimal" carrier frequency, maximum reliable range of seventeen meters can be expected. Of course, the use of very long wavelengths could substantially increase this range, but antenna dimensions would quickly become totally unmanageable for the free swimming diver. The primary problem inherent in this form of propagation is the very small distance needed to achieve skin depth; sea water is simply a very poor dielectric, and a rather good conductor.

It is well recognized that a "window" in the light attenuation characteristics of sea water exists at blue-green wavelengths. A great deal of research has been conducted into the possible use of this physical characteristic of sea water to convey data by transmission of light energy using laser technology. However, even a passing familiarity with Navy diving operations indicates that most harbors are far from translucent. It is apparent that this means of energy transmission in unsuitable for the proposed diver communications system.

Acoustic transmission, then, has been decided upon by almost all designers of underwater communications systems. Relatively low power systems can be shown to be feasible, and near-omnidirectional transmission is well within the state of the art.

Unfortunately, a very wideband acoustic signal is not practical for a number of reasons. Since low power is a

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[9]Booth, N.O., Diver Electromagnetic Communications, Naval Undersea Research and Development Center, San Diego, Ca., 1962.

criterion, operation of transducers near resonance is indicated. It is a physical characteristic of most transducers that broad-banding is not possible; mechanical resonance of these transducers is almost always characterized by relatively high Q (narrow bandpass). Furthermore, attenuation of the acoustic wave due to absorption, scattering, and other effects is far more severe at higher frequencies than low frequencies.

Attenuation is one of the most critical problems faced by the designer of underwater acoustic transmission equipment. A number of investigators[10] have measured and predicted attenuation losses for various frequencies and several differing expressions for attenuation can be found in the literature. A rather thorough discussion of the theoretical prediction of attenuation losses in sea water is provided by Urick.[11] The theory expressed accurately predicts the nature of attenuation losses as experimentally verified by tens of thousands of measurements made at sea by Shulkein and Marsh.[12] The modified form of attenuation prediction is given by:

$$\alpha = 7.2 \times 10^{-3} f^2 \text{ dB/KM}$$
$$(f \text{ in kHz})$$

For example, to operate a wideband signal on a carrier frequency of 100 kHz with an overall bandwidth of 100 kHz,

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[10] NAVSHIPS 0967-129-3010 Introduction to SONAR Technology, p. 25, Bureau of Ships, Washington, D.C., December 1965.

[11] Urick, Robert J., Principles of Underwater Sound for Engineers, p. 87-90, McGraw-Hill, 1967.

[12] ibid., p. 88.

the amplitude of the received signal at the highest frequency would differ from that of the lowest by 136 dB! Even using a system which does not intentionally modulate the amplitude, a limiter system with dynamic range of over 100 dB is not considered feasible.

Since acoustic transmission has been decided upon for the system, a brief investigation into the nature of underwater noise is necessary. "Ambient noise" may be said to be noise inherent in the sea itself. It is the background noise level encountered in the absence of an identifiable radiating source. A great many studies of ambient noise have been conducted recently; measurements over the range 1 Hertz - 100 kHz indicate that certain consistent characteristics have been measured in readily identified frequency bands. Figure 4 shows the noise power spectral density over this frequency range.

Ocean turbulence, in the form of random currents, is capable of creating a significant noise background. This is the major source of ambient noise in the 1 - 10 Hz spectrum. The magnitude of the spectral density of ambient noise amplitudes in this range of frequencies can be as high as 50 dB (re 1μ bar per Hz).

In the next band of frequencies (10 to 500 Hz), ship traffic and other man-made machinery are the major sources of ambient noise. This radiation source is often found to be relatively independent of weather and wind conditions. Typical average spectral noise density at these frequencies is on the order of -30 dB (re 1μ bar per Hz).

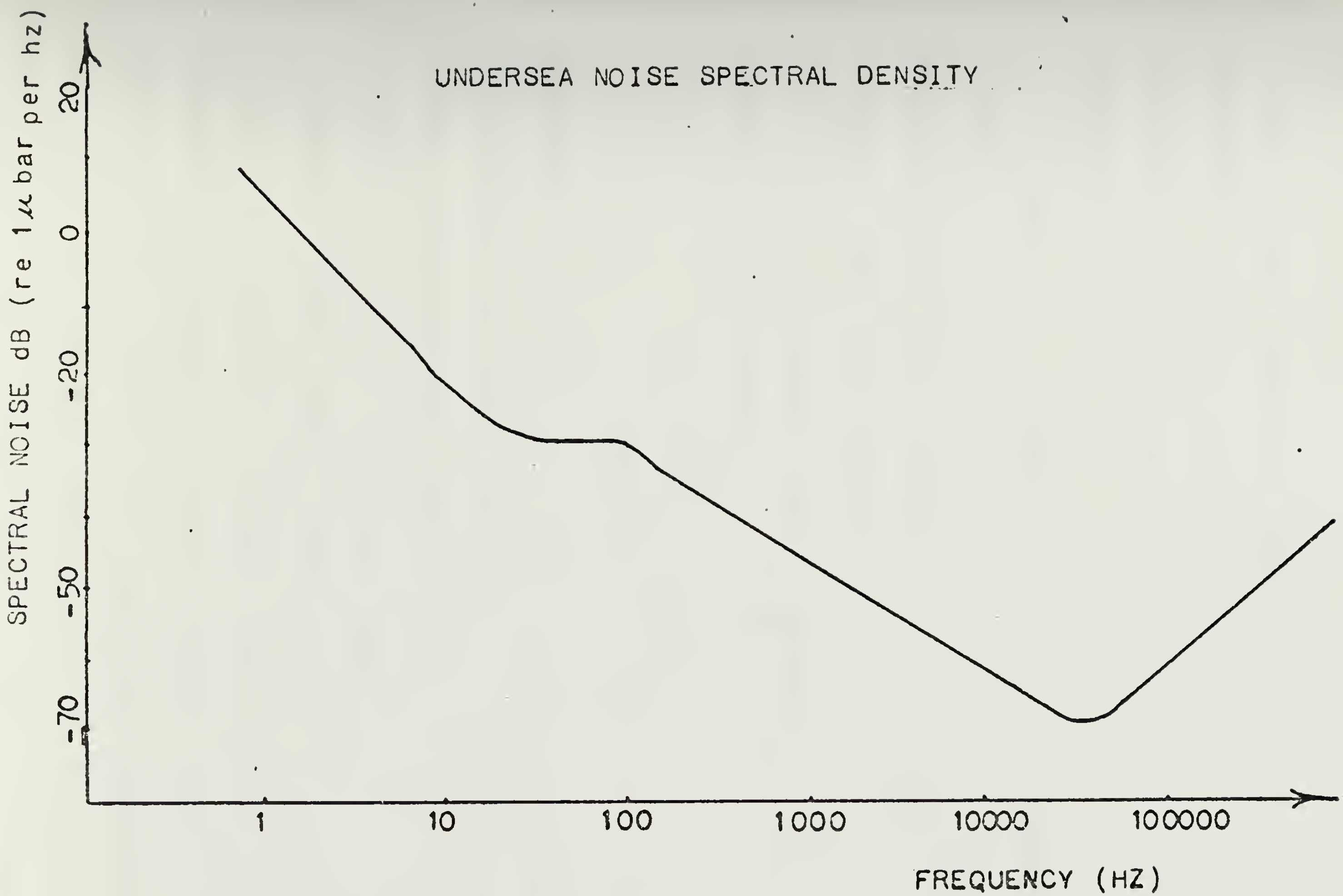
UNDERSEA NOISE SPECTRAL DENSITY
SPECTRAL NOISE dB (re 1 μ bar per hz)
20
0
-20
-50
-70
1
10
100
1000
10000
100000
FREQUENCY (HZ)

FIGURE 4

At still higher frequencies, ambient noise is primarily attributed to weather conditions. Wind noise, surface wave action and local turbulence make up the majority of this noise. The well-known Knudsen Spectra, reproduced here as Figure 5, clearly show a phenomenon that has been repeatedly measured at various undersea sites all over the world: noise in the 500 hertz to 100 kHz spectrum is directly related to the local weather conditions.

Thermal noise, usually considered the dominant noise source in radio and radar systems (which operate in air), is usually considered virtually insignificant at frequencies below 50 kHz. In fact, it has been shown that a perfectly efficient non-directional hydrophone receives equivalent noise spectral level which can be described by:

$$NL = -115 + 20 \log f$$

where f is in kHz.

At 40 kHz, the noise level due to thermal activity is, thus, about -83 dB (re 1μ bar per hertz). Even at Sea State 0, (glassy sea and no wind) Knudsen Noise Level has been measured at around -78 dB (re 1μ bar per hertz).

Forty kHz was chosen as the carrier frequency for this project for a number of reasons, such as availability of transducers, and ease of circuit design. However, the major deciding factor was the noise considerations discussed above. A definite "dip" in the spectral noise curve occurs consistently near 40 kHz. Even though it might be argued that acoustic energy attenuation might be considerably less severe

KNUDSEN CURVES

AMBIENT NOISE LEVELS IN SEA WATER

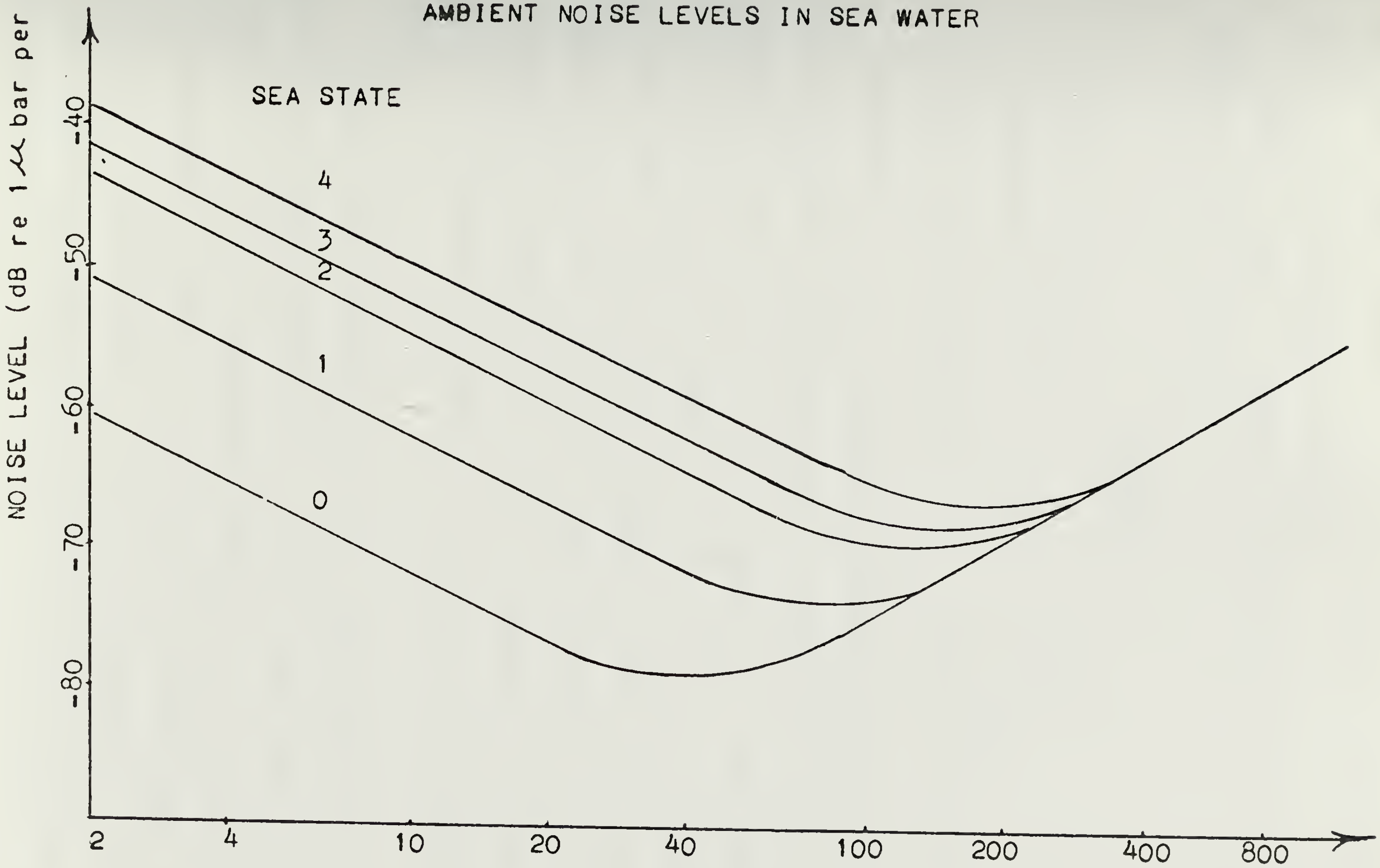

FIGURE 5

at frequencies much lower, it can easily be seen on Figure 4 that the price one has to pay is rapidly increasing noise level with decreasing frequency.

Table I shows a summary of factors encountered by the 40 kHz carrier frequency.

TABLE I

Carrier frequency	:	40 kHz
Attenuation	:	11.5 dB/KM
Noise spectral density. .	:	-69 dB (re 1μ bar per Hz)

Returning to the tradeoff between reduction of attenuation with decreasing frequency which is concurrent with significant increase in noise power spectral density, a more quantitative justification for the choice of a 40 kHz carrier frequency can now be made. By way of an example, it will be assumed that compatibility with other U.S. Navy underwater communication equipment is desired and a carrier frequency of about 8 kHz is to be considered because of its much lower attenuation losses.

Assume that a transmitter with source level SL operates at the two given frequencies: 40 kHz and 8 kHz. In both cases its energy is received R KM away in a given direction. The receiver's bandpass is B Hz and its performance characteristics are exactly equal at the two frequencies. The following table summarizes comparative performance of the system at the two frequencies:

TABLE II

	8 kHz	40 kHz
Transmitted Power	SL	SL
Attenuation	.46 R dB	11.5 R dB
Received Signal Power	SL - .46 R dB - 20 $\log_{10}$ R	SL - 11.5 R dB - 20 $\log_{10}$ R
Received Noise Power	-12 + 20 $\log_{10}$ B	-78 + 20 $\log_{10}$ B
Signal to Noise Ratio	SL - .46 R + 12 - 20 $\log_{10}$ B - 20 $\log_{10}$ R	SL - 11.5 R + 78 - 20 $\log_{10}$ B - 20 $\log_{10}$ R

Solving for that range beyond which the S/N of the 8 kHz signal exceeds that of the 40 kHz carrier:

$$SL - .46\,R + 12 - 20\log_{10} B - 20\log_{10} R$$

$$> SL - 11.5\,R + 78 - 20\log_{10} B - 20\log_{10} R$$

$$R > 5.98\ \text{KM.}$$

Thus, in order to maximize S/N at the short range system's receiver, 40 kHz is seen as the optimal frequency choice.

One final characteristic of the ocean environment of great concern is that of multipath radiation. Because of the very slow nature of acoustic propagation in sea water (see Figure 6), a carrier frequency of 40 kHz has a wavelength of only 3.1 cm. The sea-air interface usually has a very high coefficient of reflection (typically 0.99), and depending on bottom composition and the presence of objects such as air cylinders of nearby divers, or ships' hulls, a pulse of energy

SOUND VELOCITY AT ZERO DEPTH
(SALINITY: 35ppt)

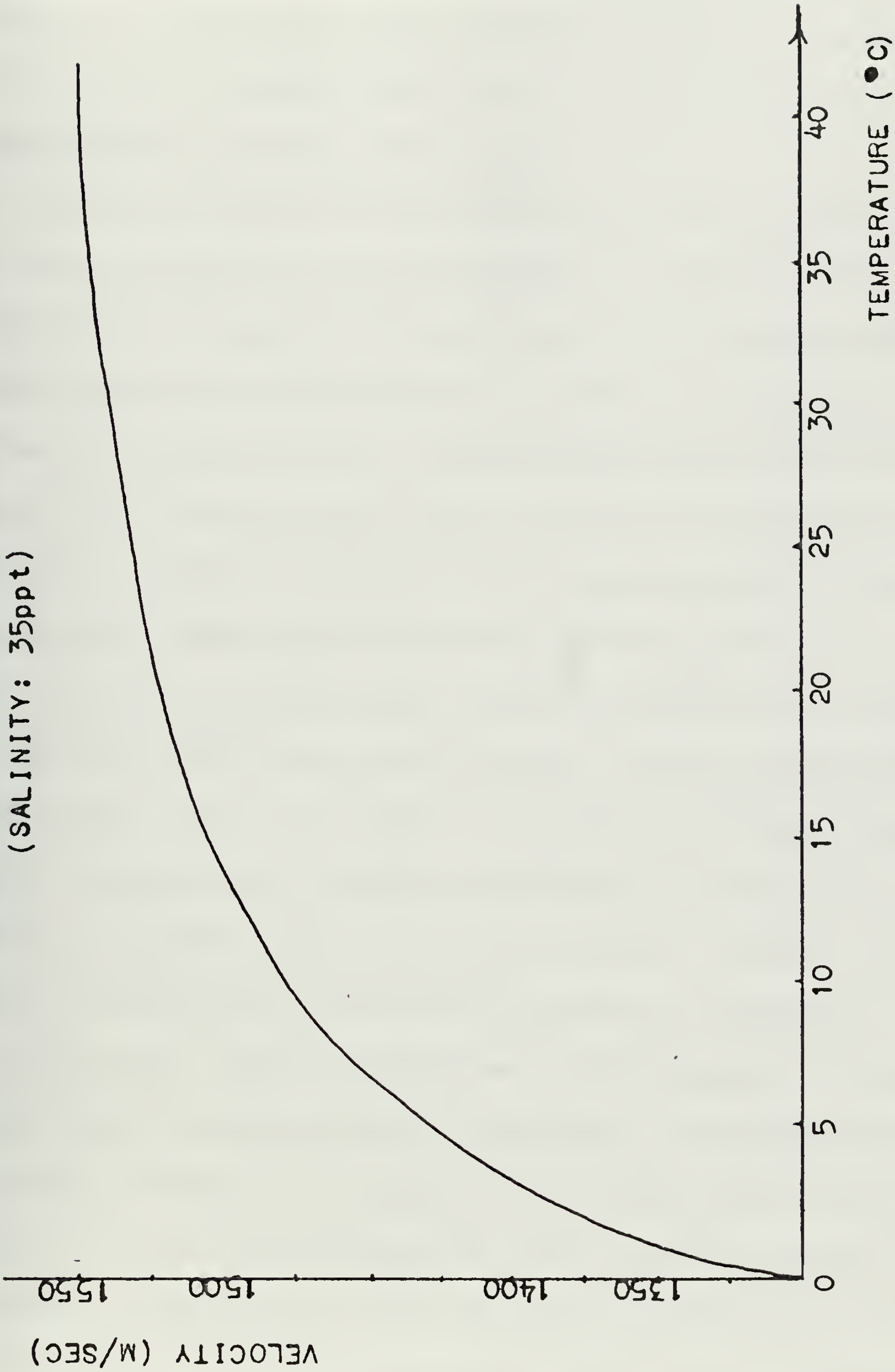

FIGURE 6

can be received many times with almost no difference in amplitude. Due to the very short wavelengths involved, rapid fading and the destructive interference due to a very slight motion such as gentle wave action, could severely distort the amplitude of the received signals.

In the selection of a modulation mode, virtually any form of pulse modulation would probably be almost impossible. A single pulse could be received several times by multipath, and could easily be demodulated as several separate pulses, thus eliminating from further consideration pulse position modulation, pulse coded modulation, and any of the delta modulation schemes. Since a previously-sent pulse could be received at nearly the same amplitude as the direct path pulse, thus modulating its width, pulse duration modulation would almost certainly suffer severe distortion. Pulse amplitude modulation (PAM) would, in reality, be little different from AM-DSB-WC modulation; one representation of sea water as the acoustic medium is as a low pass filter. Abrupt, vertical edges to the pulses (infinite frequency components) could not be preserved. PAM, in addition to multipath interference, would have a second serious drawback: attenuation differences between low and high frequency components would certainly greatly alter the amplitude of the received signal, and obviously, unlike other forms of pulse modulation, hard limiting would merely strip the received signal of its information content.

One means to achieve some form of pulse modulation deserves serious future consideration. If a series of pulses is

transmitted on two separate carrier frequencies simultaneously, some correlation scheme at the receiver could be utilized to extract only the direct path and ignore all pulses received during the time when the multipath pulses could arrive. The author has previously designed and built a pulse position modulation modulator and demodulator which is time-division-multiplexed to allow three or more separate messages to be transmitted simultaneously. This system could conceivably allow for continuous monitoring of some biological functions while allowing plain voice communications to continue unaffected. A future project might be the incorporation of a frequency diversity technique, such as that explained above, to take advantage of the superior performance of pulse modulation in a high noise environment.

It follows, then, that some sort of analog modulation is required to enable the acoustic carrier to be modulated with the message. The carrier's amplitude, phase, or frequency could be varied in accordance with the applied message. Since it is not the intention of this paper to publish an in-depth study of the subject of communication theory, the following discussion of amplitude modulation (AM) and angle modulation, including frequency modulation (FM) and phase modulation (PM) will be necessarily brief.

Since any hydrophone utilized by the receiver has a sensitivity characteristic, ML, which relates input pressure to voltage applied to the receiver, the familiar units of volts and watts will be utilized in the discussion rather

than pressure intensity and power. The comparison of the various forms of modulation which follows summarizes those aspects of each form of modulation which are pertinent to diver communications. The following symbology is used in this discussion:

- $v_c(t)$: the time description of the transmitted voltage
- $v_c(f)$: the frequency description of the transmitted voltage
- $m(t)$: the modulating voltage; the message
- $M(f)$: the frequency description of the message
- f_m : the highest modulating frequency
- f_c : carrier frequency
- δ : the Duroc delta function
- A_c : a constant

Figure 7 depicts a typical modulating voltage in both the time and frequency domains. Figure 8 shows the frequency description of the transmitted signal for all the AM forms of modulation. Table III summarizes some of the characteristics of these forms of modulation.

Additional observations follow: (1) In all three cases the S/N of the recovered message is, at best, only as good as that of the received signal. (2) All three are forms of linear modulation; that is, the amplitude of $v_c(t)$ varies directly with the amplitude of $m(t)$. (3) For a given total transmitter power at a given receiver range, the S/N received by AM-DSB-S/C modulation will be exactly the same as AM-SSB-S/C and will be four times that of AM-DSB-W/C.

THE MODULATION VOLTAGE

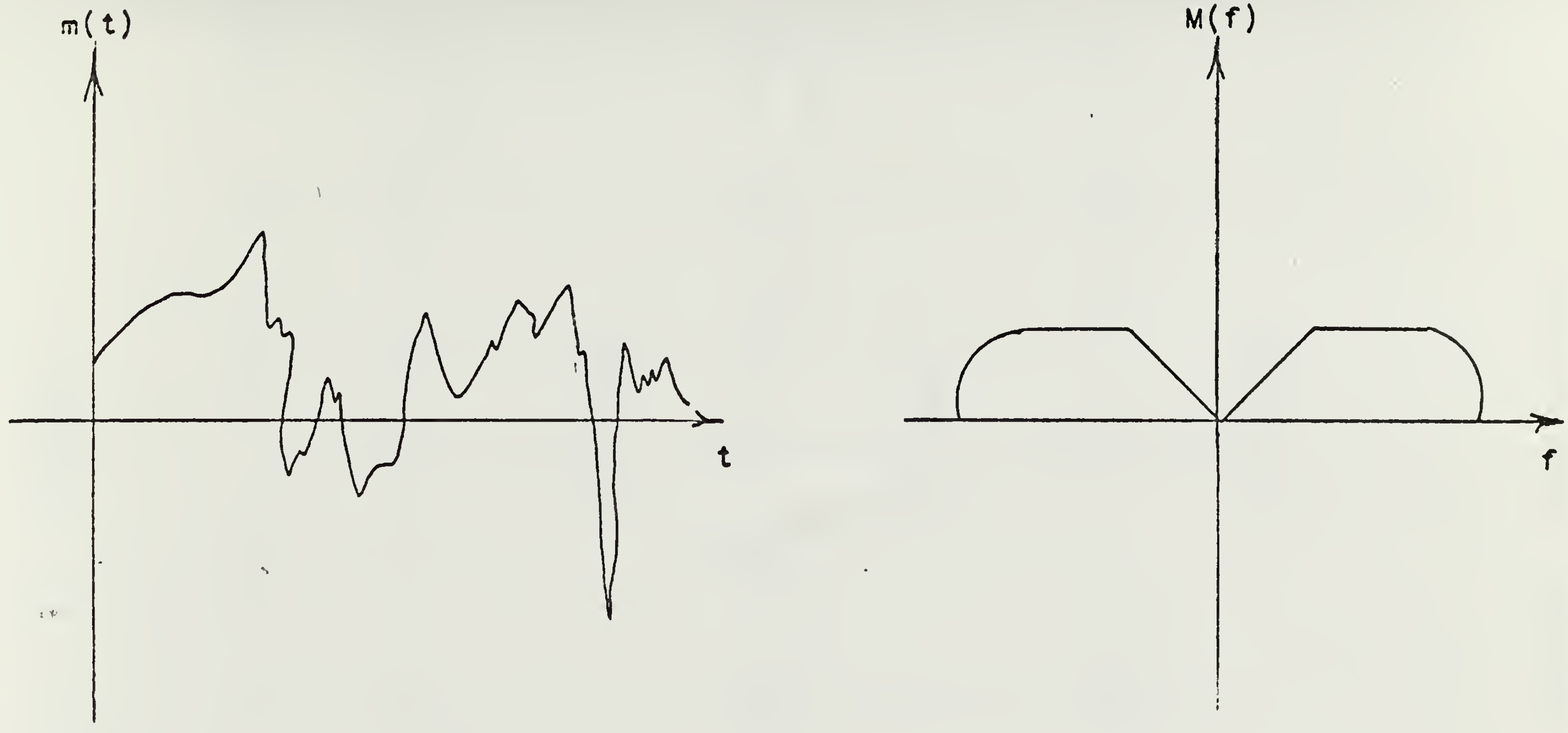

FIGURE 7

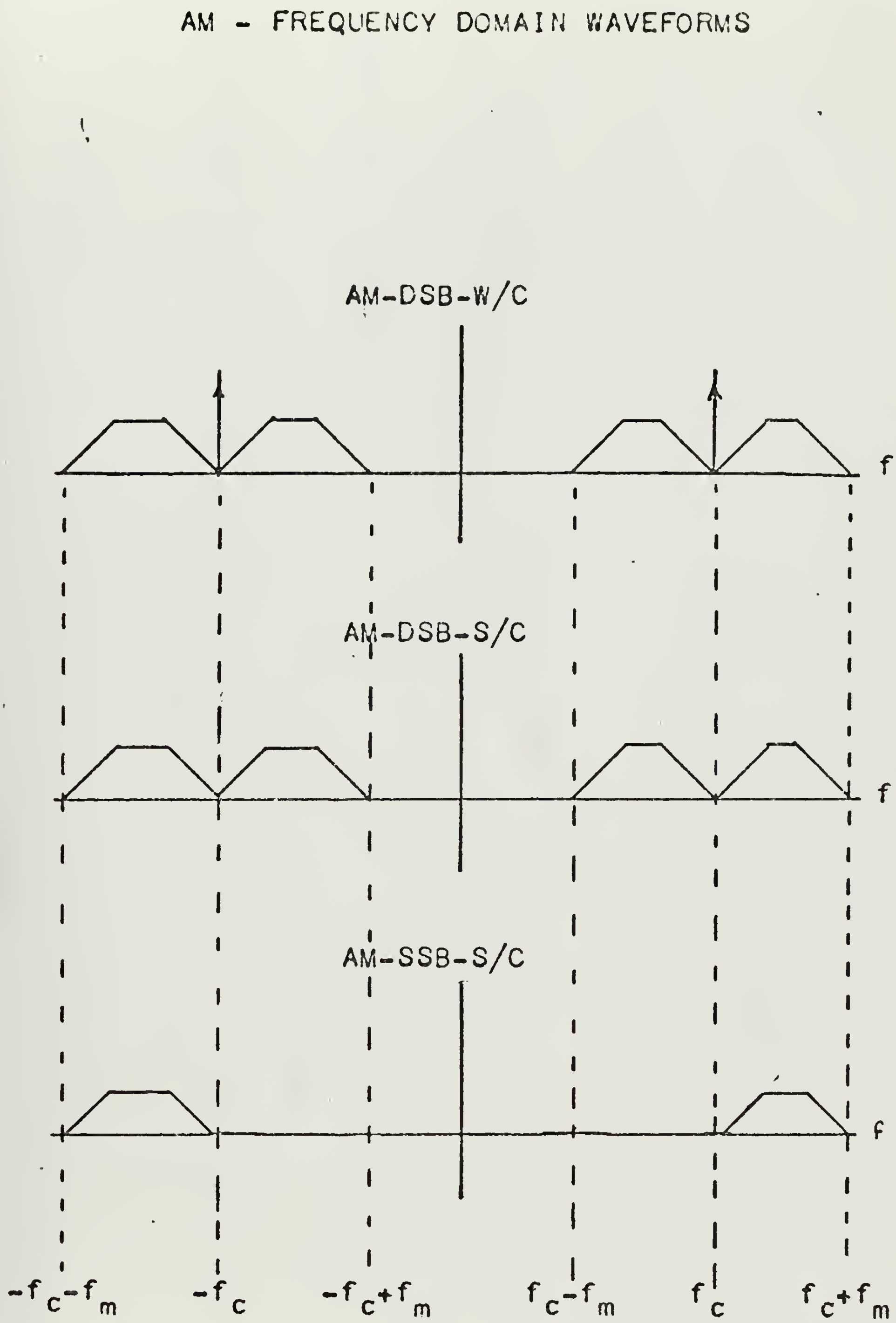
AM - FREQUENCY DOMAIN WAVEFORMS
AM-DSB-W/C
AM-DSB-S/C
AM-SSB-S/C
f
f
f
$-f_c-f_m$
$-f_c$
$-f_c+f_m$
f_c-f_m
f_c
f_c+f_m

FIGURE 8

TABLE III

	AM-DSB-W/C	AM-DSB-S/C	AM-SSB-S/C
v_c (t)	$A_c[1 + m(t)] \cos 2\pi f_c t$	$A_c\ m(t) \cos 2\pi f_c t$	--
V_c (f)	$\frac{1}{2}A_c[\delta(f + f_c) + \delta(f - f_c)$ $+ M(f + f_c)$ $+ M(f - f_c)]$	$\frac{1}{2}A_c M(f) * [\delta(f + f_c)$ $+ \delta(f - f_c)]$	$\frac{1}{2}A_c M(f) * \delta(f + f_c)$ or $\frac{1}{2}A_c M(f) * \delta(f - f_c)$
Bandwidth	$2f_m$	$2f_m$	f_m
m(t) = 0	carrier still present	no carrier	no carrier
Portion of Transmitter Power Consumed by Carrier	67%	0	0
S/N Necessary for Good Demodulation	20 dB	20 dB	20 dB

AM-DSB-W/C is the easiest modulation mode to implement in circuitry and tuning the receiver for understandable demodulation is not critical. However, its disadvantages shown above and in Table III far outweigh these two advantages. Because of the wide bandwidth and the percentage of power wasted by the carrier which conveys no information, this form of modulation is eliminated from further consideration.

Although, as stated above, AM-DSB-S/C occupies twice the bandwidth of AM-SSB-S/C, and thus opens the receiver to twice as much noise power, the information, carried by both sidebands, provides effectively twice the signal power. Thus, their performance could be seen as identical unless one further consideration is made. Coherent (the receiver's local oscillator must be in phase with the transmitter oscillator) demodulation is necessary for proper demodulation of the AM-DSB-S/C signal. With the use of a Costas Loop or some other phase-locking scheme, this is possible but fails to provide any advantage over the use of AM-SSB-S/C. Therefore, single sideband is seen as the most advantageous form of AM to be considered for implementation in the diver communications system. Any further comparisons or references to AM made below will be with respect to single sideband.

Several of the engineers who are responsible for the design and testing of the Navy diving communications systems today, seem to be in general agreement that some form of AM, preferably single sideband, places the greatest amount of signal power in the smallest bandwidth, and hence would be

the best approach to the underwater communication problem. The arguments for remaining with narrow band signals are very much the same as those faced by Armstrong when he advocated a spreading of the spectrum of the transmitted signal to improve its characteristics in the presence of a noisy environment.[13] It seemed to his adversaries totally contradictory to develop a receiver with a wide bandpass to improve the signal to noise at its output.

Primarily because of the inability to construct a truly wide band transducer which provides good sensitivity near resonance, a really wideband form of modulation is not presently feasible. It will be shown below that a "narrow band" or "medium band" form of angle modulation might be the better approach. Even though these modulation modes require the opening of the bandpass of the receiver amplifier circuits prior to demodulation, and they allow a large increase of noise power to be applied to the demodulator, the processing gain afforded by the demodulator can provide more than enough improvement to offset the increase in signal noise power delivered to the demodulator input. See Figures 9 and 10.

Frequency modulation, unlike the various forms of AM, is <u>not</u> a linear form of modulation. That is to say, its spectrum description might bear no resemblance to the modulating waveform. Here, the carrier waveform can be described by:

---

[13]Erickson, D. V., <u>Armstrong's Fight for FM Broadcasting</u>, p. 20-86, University of Alabama Press, 1968.

$vc\ (t) = Ac \cos\ (2\ f_c t + \theta)$ θ is constant

Let $\phi = 2\pi f_c t + Kpm\ (t)$ K_p-units: radians/volt

Thus, a description of phase modulation would be:

$$Vc\ (t) = Ac \cos\ [2\pi f_c t + K_p M\ (t)]$$

Likewise, the FM waveform could be described by:

$$Vc\ (t) = Ac \cos\ [2\pi f_c t + 2\pi K_f \int m\ \dot{(t)}\ dt]$$

K_f units : h_3/volt

$\Delta f = [K_f \int m(t)\ dt]$ max : maximum excursion from f_c

Let $\beta = \dfrac{\Delta f}{f_m}$: deviation ratio

Expanding the description for $V_c(t)$ shown above:

$$\begin{aligned} V_c(t) = J_o(\beta) \cos \omega_c t &+ J_1(\beta) \cos\ (\omega_c + \omega_m)t \\ &+ J_1(\beta) \cos\ (\omega_c + \omega_m)t \\ &+ J_2(\beta) \cos\ (\omega_c + 2\omega_m)t \\ &+ J_2(\beta) \cos\ (\omega_c - 2\omega_m)t \\ &+ J_3(\beta) \cos\ (\omega_c + 3\omega_m)t \\ &- J_3(\beta) \cos\ (\omega_c - 3\omega_m)t \\ &+ \\ &\cdot \\ &\cdot \\ &\cdot \\ &+ J_n(\beta) \cos\ (\omega_c + N\omega_m)t \\ &+ (-1)^N J_N(\beta) \cos\ (\omega_c - N\omega_m)t \end{aligned}$$

where $J_n(\beta)$ is a Bessel function of the first kind.

If narrow band FM is defined as the first line of the a above expansion, it can be seen that narrow band FM, consisting only of a carrier, one upper and one lower sideband, is exactly synonomous with AM-DSB-WC with the one difference being the phase of the sideband is in quadrature with the phase of its carrier. Thus, except for the lower S/N required for its demodulation (to be discussed below) little has been gained over the AM case, until processing gain is considered.

Carson[14] published the following practical description of the bandwidth of a given FM signal:

$$BW \simeq 2f_{max} \ (\beta + 1)$$

An achievable Q for transducers in the region of the selected carrier frequency is about 2. Thus a peak deviation of 10 kHz either side of a 40 kHz carrier appears to be possible. Since f_m = 7.5 kHz, β is less than 2.

A comparison of this medium band FM signal with AM-SSB-SC follows. Signal to noise ratio improvement due to processing gain in the demodulation phase of FM transmission is shown in Figure 9. Assuming S/N is excess of threshold (see Figure 10) for a given demodulation, with β = 2, FM provides approximately 15 dB improvement. As shown in Figure 10, very low input S/N ratio can be accommodated by the use of a phase locked loop in the demodulator. Thus, the design parameter will include the use of a phase locked loop.

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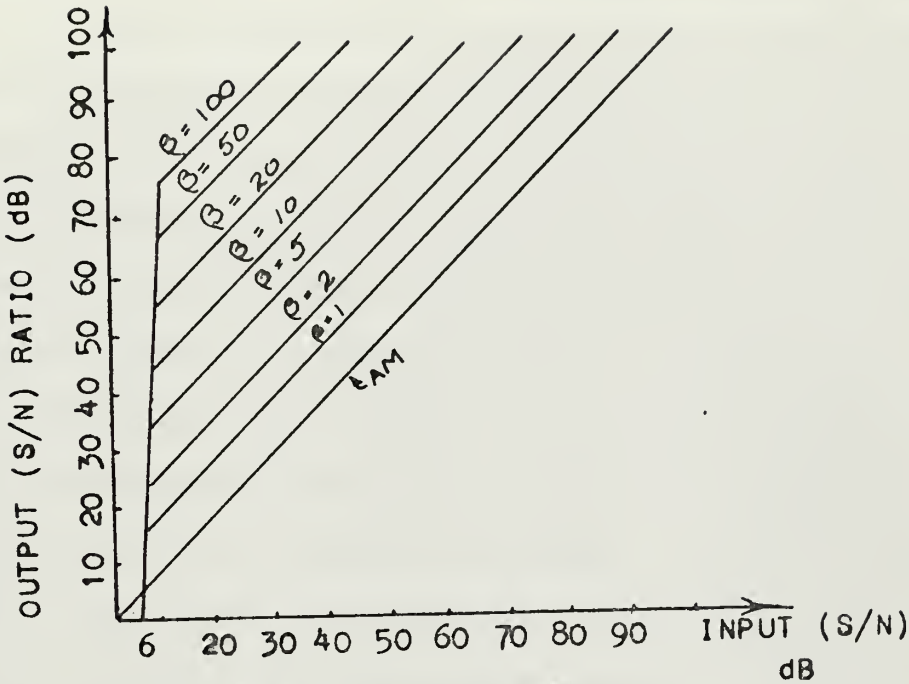
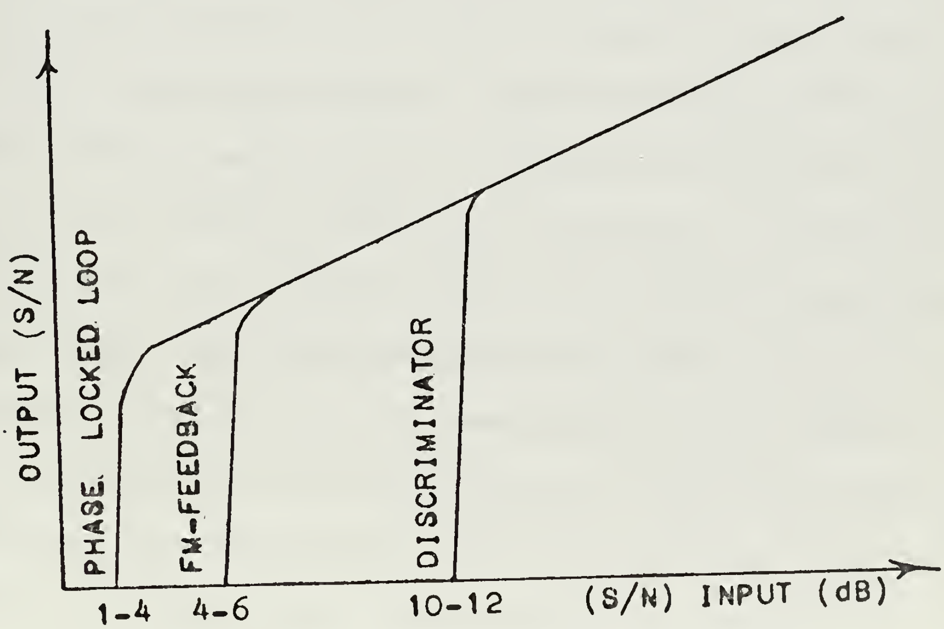
[14] Hund, August, Frequency Modulation, p. 78, (McGraw Hill Book Co.), 1948.



PROCESSING GAIN - FM DEMODULATION

FIGURE 9



THRESHOLD FOR VARIOUS DEMODULATORS

FIGURE 10

Performance in the multi-path environment appears to favor FM as well.[15] The AM signal can fairly easily be made unintelligible by multipath, whereas harmonic distortion of the audio components will be the major result of FM multipath.

The use of FM would provide significant improvement in signal to noise over the use of any AM system, provided that threshold is exceeded, as shown by Figure 9. Assuming it is possible to construct a hard limiter with dynamic range of about 20 dB, amplitude distortion caused by the underwater acoustic transmission is reduced to minimal levels of audio distortion at the output in FM, whenever the demodulator's threshold is exceeded. With the recent advent of integrated circuit form phase locked loop, threshold can be reduced to as low as 1 dB S/N input levels. It is for these reasons that underwater diving communications using medium band FM and phase locked loop demodulation would provide significant improvement over existing systems. Even narrow band FM which, at the very least, could contain sidebands that extend the bandwidth to twice that of SSB, would provide a significant advantage. Due to the capture effect and the low pass nature of the audio amplifier and the listener's ears, as well as the processing gain realized in its demodulation, the quality of the audio presented to the diver would be relatively noise-free and undistorted and much lower S/N input would be required for the receiver to provide good demodulation.

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[15] Reference Data for Engineers, p. 21-25, International Telephone and Telegraph Corp., 1973.

It has been pointed out above that the underwater environment provides two major obstacles to the transmission of a message by modulating a carrier: high attenuation to signal power and high ambient noise power. Thus, a high signal to noise ratio is difficult to achieve. In order to deal with this problem and others such as multipath and fading, the optimal approach to the transmission of the message is seen as the transmission of a 40 kHz acoustic wave frequency modulated by the message.

IV. ULTRASONIC TRANSDUCER: DESIGN, CONSTRUCTION, AND EVALUATION

A. CHOICE OF TRANSDUCER TYPE

Since it was decided that the ultrasonic acoustic wave is the desired medium of transmission, a device for conversion of electrical energy to sound pressure is necessary. This transducer must be of the variety for which reciprocity principles pertain. That is, the device must also be capable of operation as a receiving hydrophone in converting sound pressure to voltage levels suitable for processing in the receiver. Since the system is to be compact, lightweight, and inexpensive, operation near resonance is indicated; by providing the best possible sensitivity at the transducer, power output requirements in the transmitter and voltage amplification requirements in the receiver are minimized.

After some investigation into the various types of transducer element commercially available, a hollow cylindrical piezo-electric element was selected such that, working in the radial mode, essentially omni-directioral coverage is obtained in one plane. The element, provided by International Transducer Corporation, was mounted in a brass enclosure which also provides a suitable housing for a torridal inductor which is used for tuning.

B. TRANSDUCER MOUNTING

The physical arrangement of the transducer mounting enclosure is shown in Figure 11. The housing is constructed of brass. "0" rings are utilized to ensure watertight integrity.

Lieutenant Commander Fred R. Crawford, USN, a Master's student in the Acoustics Engineering curriculum, provided a great deal of assistance in the testing and evaluation of the transducers as described below. The following table summarizes the physical characteristics of the piezo-electric transducer element.

TABLE IV

material	:	lead zirconate with silver surfaces bonded inside and out.
physical dimensions:		height -- 1/2" (1.27 CM) outer diameter -- 1" (2.54 CM) wall thickness -- .0075" (.019 CM)
medium	:	air (atmospheric pressure) inside, outer surfaces coated with neoprene which is in direct contact with sea water.
mode	:	radial expansion and contraction of diameter Acoustic axis is perpendicular to cylinder axis.

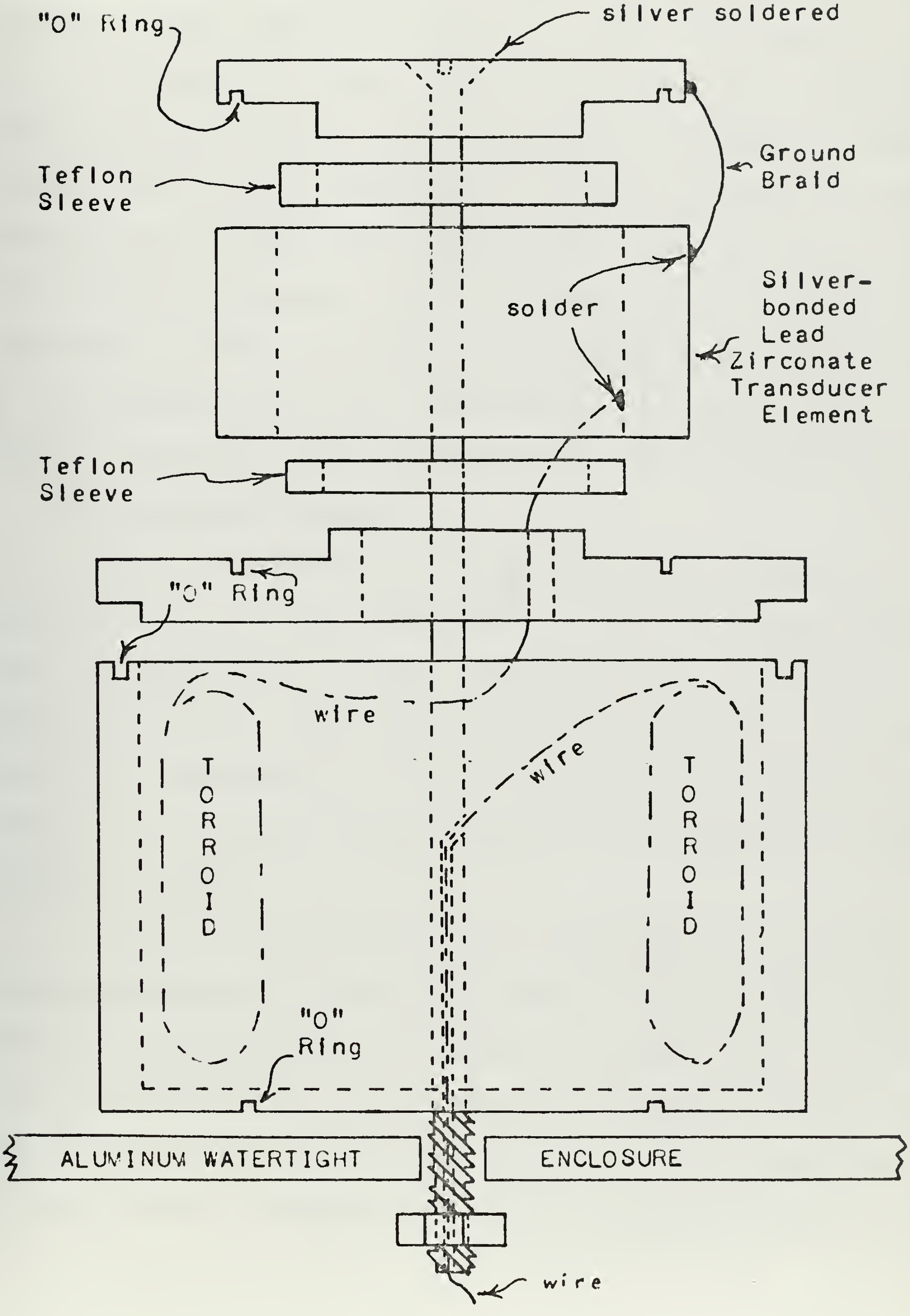
C. MEASUREMENTS

1. Electrical Impedance

Using a Drametz Engineering Laboratories model 100-B complex impedance meter with external signal generator and X-Y plotter, the transducer was immersed in a large tank of

FIGURE 11

CYLINDRICAL TRANSDUCER MOUNTING



fresh water and the resistance-reactance characteristics were obtained. Resonance occurs at slightly over 38 kHz, the impedance at that frequency being about 550 -j1200 ohms.

In order to cancel out the capacitive reactance of the transducer, a series coil was wound on a torroid form to provide about 5 mH of inductance. In this configuration the impedance presented to the transmitter's final amplifier is nearly totally resistive, and is on the order of 550 ohms in magnitude. Figure 12 is a graph of impedance characteristics of the transducer and coil combination. By reciprocity, a similar impedance should be realized in the hydrophone mode.

2. Sensitivity Levels

Two conversion factors are generally utilized to describe a bidirectional transducer: (1) *SL* denoting speaker level, or conversion of volts applied by the transmitter to pressure produced in the water, and (2) *ML* denoting microphone level, or a conversion of pressure received to voltage provided to the receiver. These factors are collectively termed sensitivity level.

Figure 13 indicates the experimental setup used in the determination of sensitivity levels. A calibrated hydrophone of the designation LC-5-2 (serial number 244) was installed in the position shown as the right hand element in the figure. This hydrophone was used as the standard for the measurements; the manufacturer's calibration curve is shown as Figure 14.

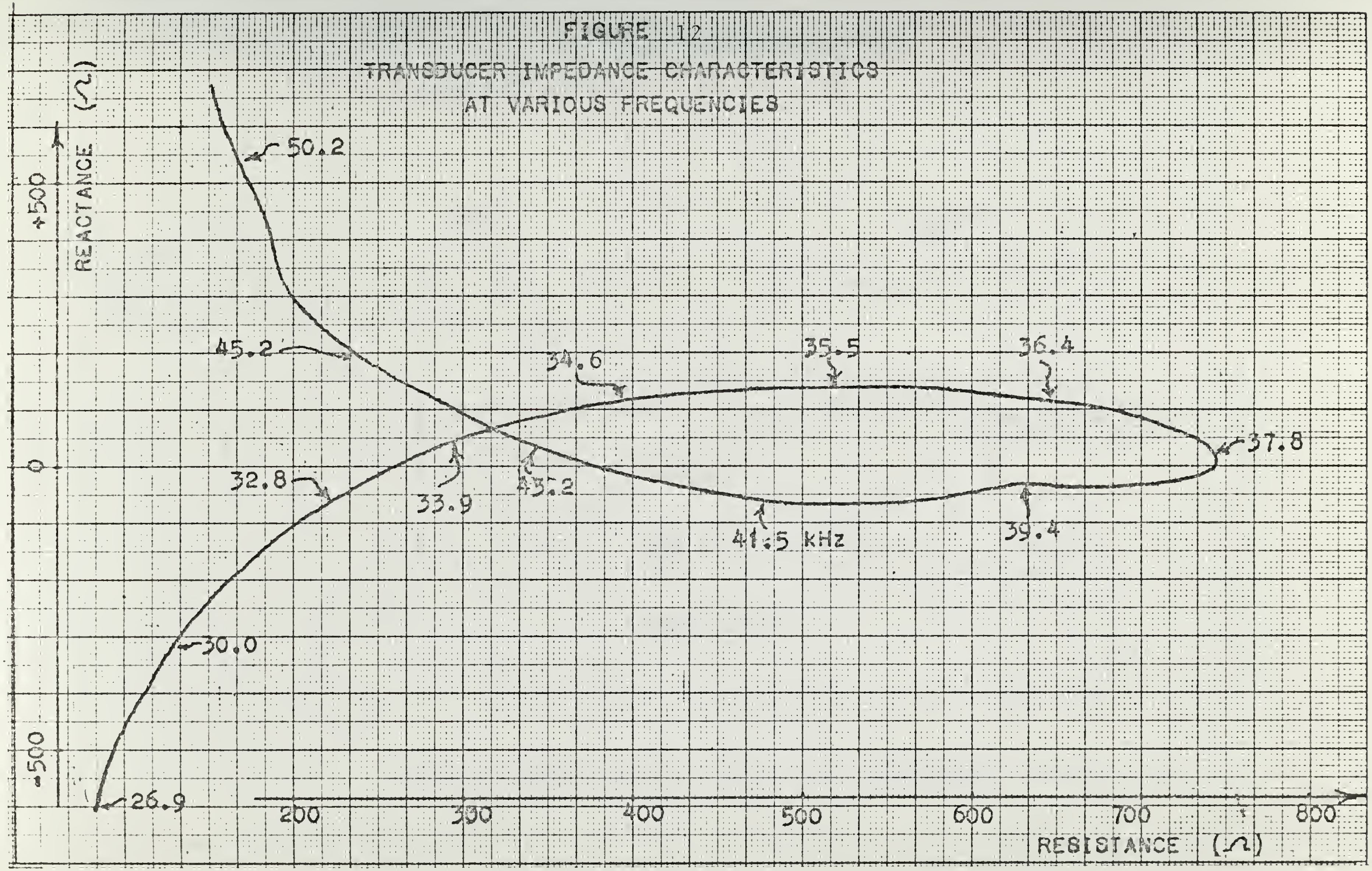


FIGURE 12
TRANSDUCER IMPEDANCE CHARACTERISTICS
AT VARIOUS FREQUENCIES

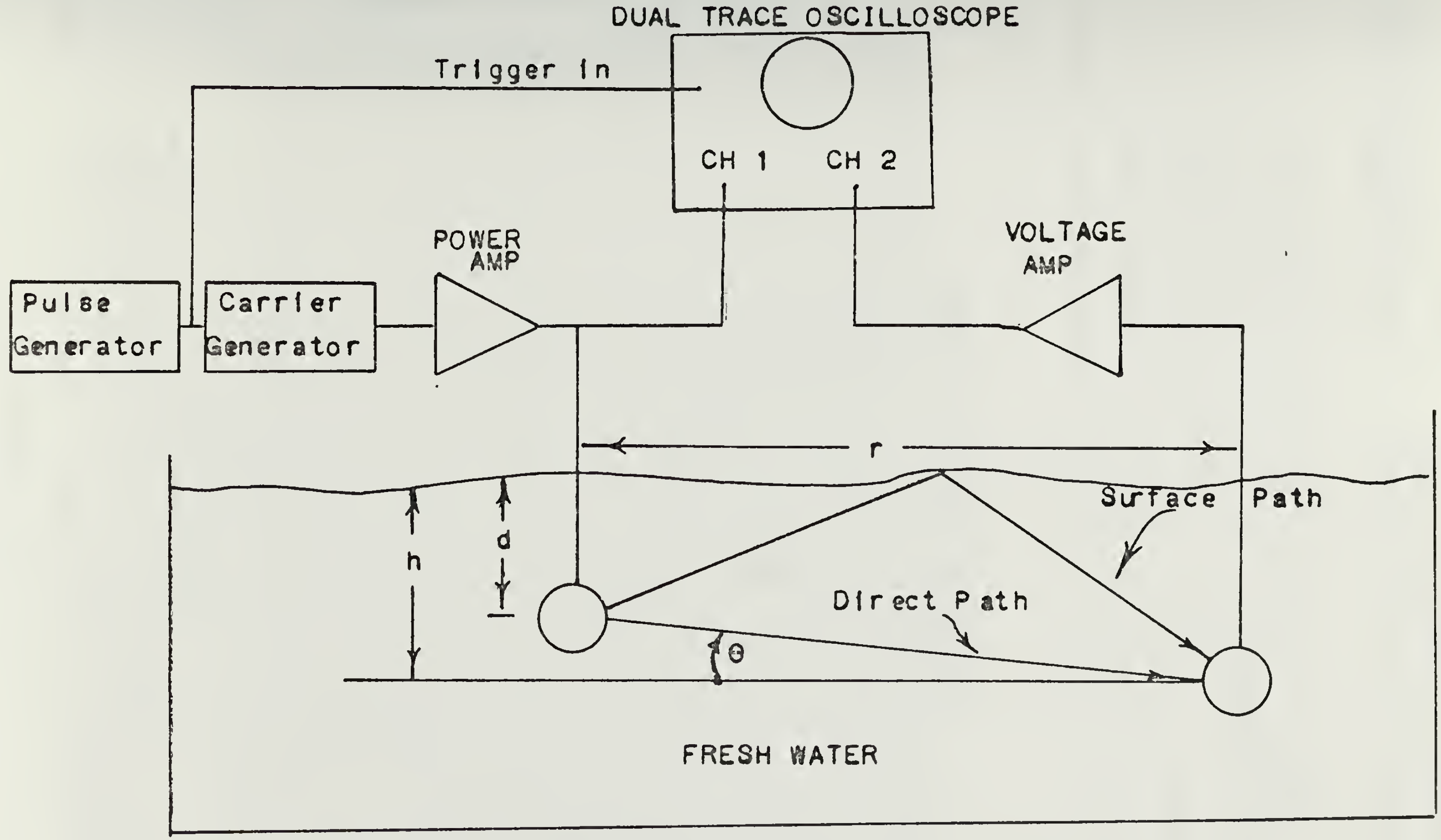


Figure 13 EXPERIMENTAL SETUP FOR TRANSDUCER CALIBRATION

environmental and industrial products division
A Division of The Susquehanna Corporation
7800 Deering Avenue • Canoga Park • Calif. 91304
Telephone: (213) 884-6860

LC 5-2 TRANSDUCER SERIAL NO. 244

AVERAGE FREE FIELD VOLTAGE SENSITIVITY -126.3 db

CABLE LENGTH 4 FT

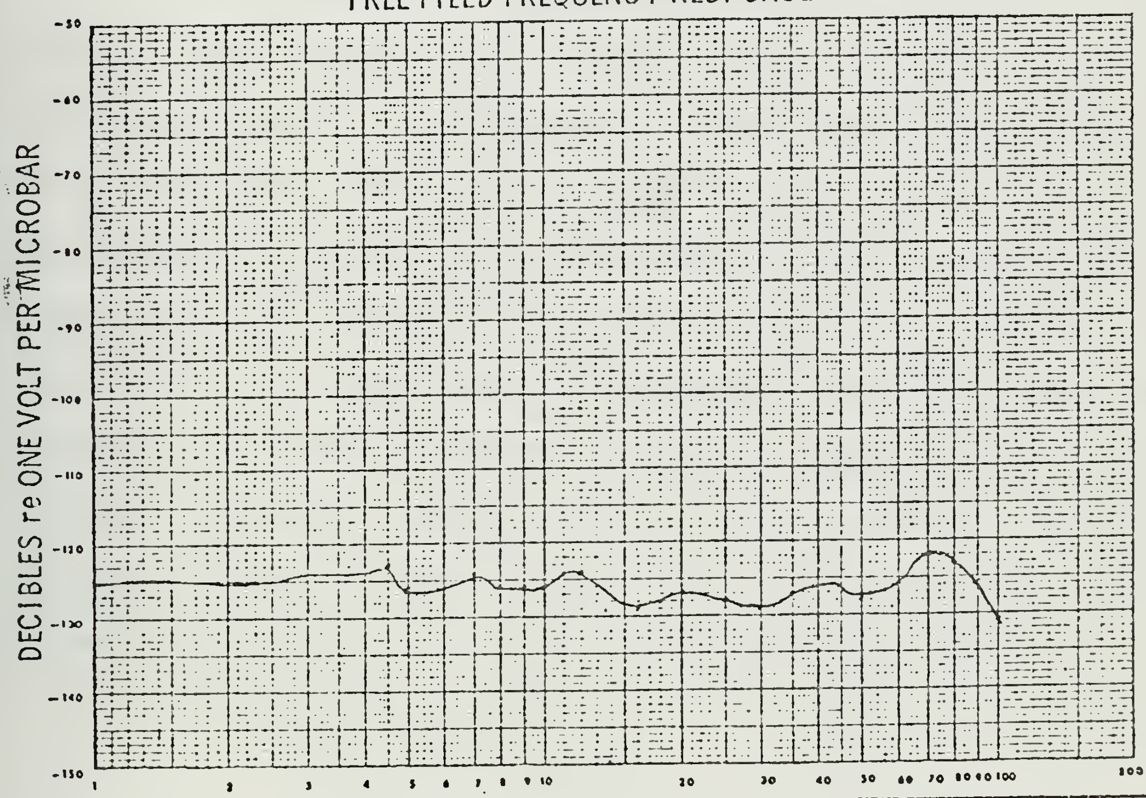
CAPACITANCE WITH CABLE 560 Pf

DC RESISTANCE 200000 MEGOHMS

MOUNTING SLEEVE LEAKAGE RESISTANCE —— MEGOHMS

DATE 5-15-73 BY ____

FREE FIELD FREQUENCY RESPONSE



METHOD OF CALIBRATION: COMPARISON

WITH STANDARD LC-32 SERIAL NO. 532

WATER TEMPERATURE 16 °C. DEPTH 10 FEET.

ATD 690401 Sheet 2

Figure 14

a. Transmit Sensitivity (Transducer Mode)

The unknown transducer was installed in the transmit mode (left-hand element in Figure 13). Depths were chosen such that d = 7.7 cm, and h = 8.3 cm. Horizontal distance, r, was then varied for each frequency so as to place the calibrated standard at the furthest maximum from the source at which a strong signal could be seen on the oscilloscope. Multipath interference was severe so great care was required to identify the actual direct path signal.

In order to convert the sound pressure levels obtained in the above manner, it was necessary to correct for the strong surface path received. A measure of direct path in an infinitely large body of water is the desired criterion for making a measure of the transducer's sensitivity level. Since, as stated above, r was varied to obtain a maximum for each frequency, the transmitting element and its virtual image located d cm above the surface appear to the hydrophone as separate radiating elements transmitting in phase. Calling the slant range from the virtual image radiator to the receiver hydrophone L, from the geometry, it can be seen that

$$L = \frac{d + h}{\sin\left[\tan^{-1}\left(\frac{d + h}{r}\right)\right]} \tag{1}$$

The total pressure received is the sum of the pressure received from the source and that from its image:

$$P_R = P_S + P_i \tag{2}$$

Since P_s is in phase with P_i, some constant K exists such that:

$$P_R = KP_S \tag{3}$$

Thus, by combining (2) and (3),

$$K = 1 + \frac{P_i}{P_S}$$

Since both sources transmit equal amplitude, the only variation in received amplitude is brought about by spherical spreading.

$$P_s = C_1 \ 20 \log r \tag{4}$$

$$P_i = C_1 \ 20 \log L \tag{5}$$

Thus,

$$K = 1 + \frac{\log L}{\log r} \tag{6}$$

And therefore, the sound pressure level (SPL) due to the direct path is:

$$SPL_s = SPL - 20 \log K \tag{7}$$

$$= SPL - 20 \log \left[1 + \frac{\log L}{\log R}\right] \tag{8}$$

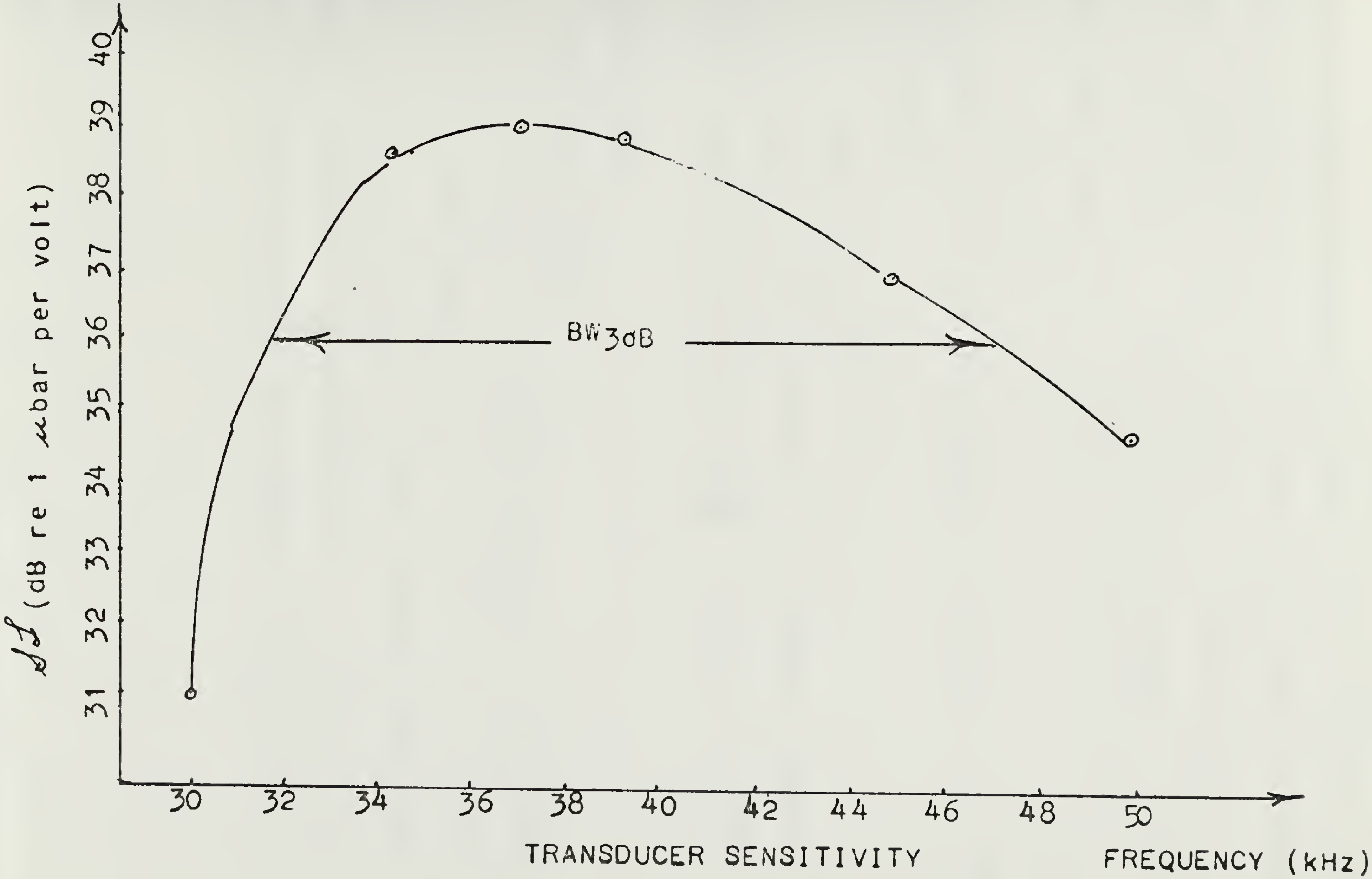
It is standard practice to refer to SPL at an imaginary sphere of radius 1 meter, as source level, SL.

$$SL = SPL \ @ \ 1 \text{ meter} = SPL_s + 20 \log \left[\frac{r}{1 \text{ meter}}\right] \tag{9}$$

The sound pressure levels determined by equation (9) were then utilized with the voltage levels (VL) at the input to the transducer to calculate the sensitivity level in the transmit mode (*SL*) at each frequency.

$$SL = SPL \ @ \ 1 \text{ meter} - VL \ (\text{re } \frac{1 \ \mu\text{bar}}{\text{volt}})$$

The sensitivity level thus determined is plotted in Figure 15.



TRANSDUCER SENSITIVITY

FIGURE 15

b. Receive Sensitivity (Hydrophone Mode)

To calibrate the transducer in the receive mode an omnidirectional transducer element was installed as in Figure 13 and the calibrated standard was used to measure pressure received at some distance, r, that provided a clean picture of the pulse. Then the voltage output of the unknown transducer was measured. The SPL was determined by:

$$SPL = VL - ML.$$

The microphone level for the unknown transducer was then calculated from:

$$ML = VL - SPL.$$

Twelve frequencies in the range 30 - 50 kHz were measured in this manner and the resulting curve is shown in Figure 16.

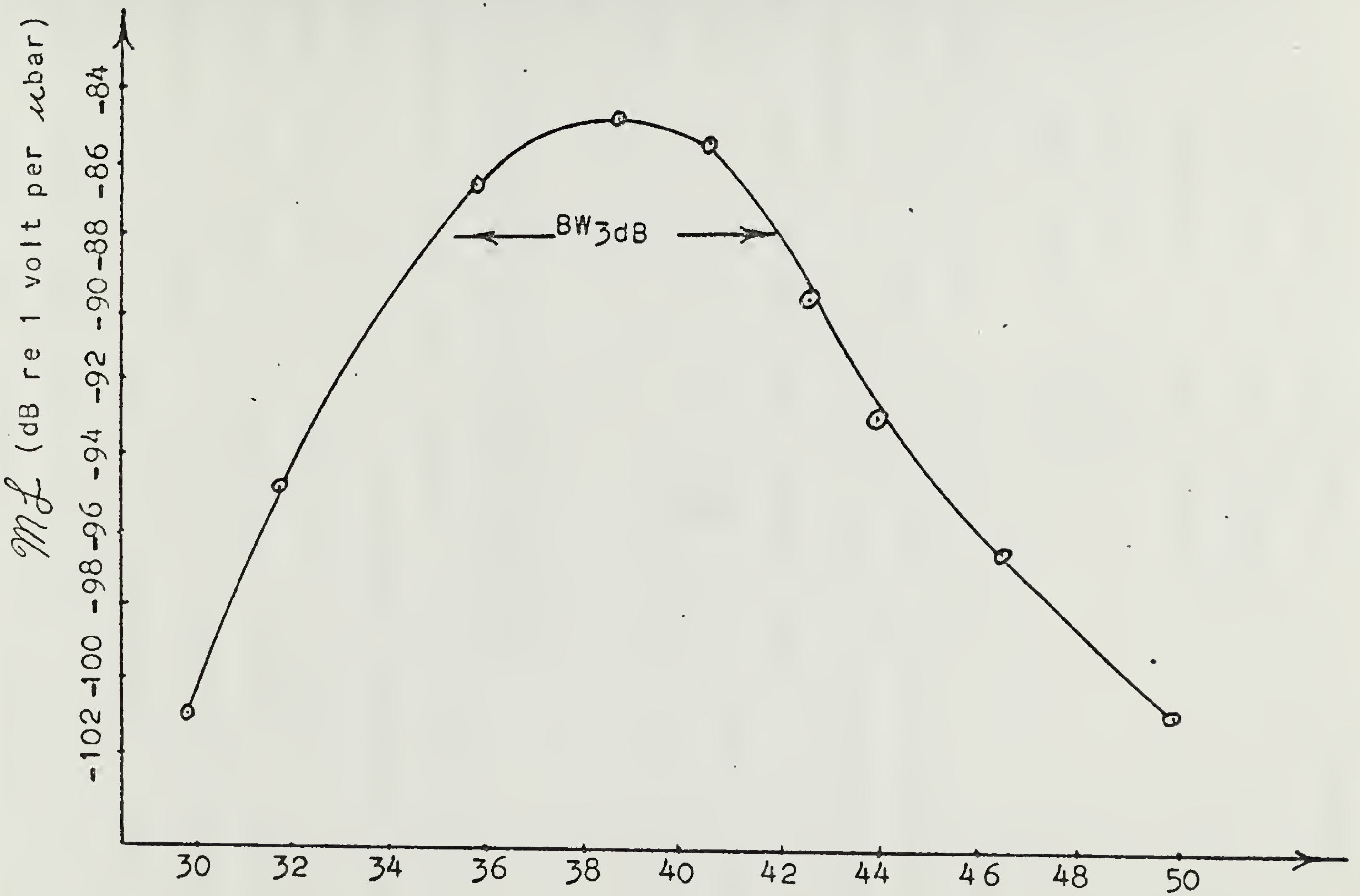
D. CALCULATIONS

1. <u>Quality Factor</u>

Determination of quality factor, Q, can be made in several ways. The most obvious of these is to study electrical impedance characteristics. Another measure of Q can be made by studying *ML* and *SL* characteristics. A third method is to measure the time constant for damping, utilizing the relationship:

$$Q = N\pi$$

where N is the number of cycles required for the oscillations to diminish the value $(\frac{1}{e})$ X starting value. The table shown below summarizes the range of values determined for the Q of the particular transducer in question.



HYDROPHONE SENSITIVITY FREQUENCY (kHz)

FIGURE 16

TABLE V

Quality Factor

Method of Determination	Q Value
1. Sensitivity level (transmit). . . .	3.3
2. Sensitivity level (receive)	6.14
3. Electrical impedance	7.15
4. Time constant method	6.28

It is therefore probable that value for Q for this transducer is somewhere in the range 3.3 to 7.15. Since operation at center frequency of 40 kHz in FM mode with maximum bandwidth of around 20 kHz was desired, a lower Q would have been desirable (near 2). However, such values for resonant transducers are thought to be difficult to attain and it was decided to proceed with the transducers.

2. System Design: Transmitter Power and Receiver Sensitivity Required

Having chosen the frequency of operation, and maximum desirable range, and, having determined transducer sensitivity levels and impedance, it is possible to calculate transmitter power level and receiver sensitivity necessary to provide good communications. In order to perform these calculations, several assumptions were made: (1) maximum sea state for conducting diving operations using DUCS-I will be sea state 2, (2) thermal noise power presented to the receiver's first stage is insignificantly small, and (3) the units yard and meter are used interchangeably. The possibility of multipath transmission is ignored in making calculations.

Table VI summarizes design criteria and calculated and measured data.

TABLE VI

Transducer Characteristics

Frequency Range	30 - 45 kHz
Average Electrical Impedance	550 + j 10 Ω
Transmitter Transformer Turns Ratio	$\frac{177}{17} = 10.4$
Impedance at Primary of Transformer	5.2 + j .09 Ω
Average Transducer *ML* . . .	36 dB (re 1 μ bar per volt)
Average Hydrophone *SL* . . .	-88 dB (re 1 volt per μ bar)

Reference to Figure 10 indicates that a conservative estimate for minimum signal to noise ratio necessary to obtain good FM demodulation utilizing the phase locked loop is 6 dB.

At the frequencies of operation, total transmission losses are made up almost entirely of spherical spreading loss (SSL) and attenuation loss, α. Spherical spreading loss is given by:

$$\begin{aligned} SSL &= 20 \log_{10} (R/1\,\text{meter}) \\ &= 20 \log_{10} (1000\ \text{meters})/1\ \text{meter} \\ &= 20\,(3) \end{aligned}$$

$$SSL_{1km} = 60\ \text{dB}$$

As previously shown, for the frequency range, attenuation losses are:

$\alpha_{30\ kHz} = 6.45$ dB/kyd

$\alpha_{45\ kHz} = 14.6$ dB/kyd

Total transmission losses at maximum range of 1 km for the worst case (highest frequency) are shown in the following equation:

$$TL = SSL + \alpha R$$

$$= 60 + 14.6$$

$$TL = 75 \text{ dB}$$

Assuming loop bandwidth of the demodulator to be 1 kHz, noise spectral density is -35 dB (re 1 μ bar).

Using the assumed minimum S/N of 6 dB, this requires signal intensity as shown:

$$S - N = 6$$

$$S = 6 + (-35)$$

$$S = -29 \text{ dB (re 1 } \mu \text{ bar)}$$

Converting this pressure to voltage level (VL_{RCVR}) presented to the receiver's "front end" --

$$VL_{RCVR} = S + ML$$

$$= -29 \text{ dB (re 1 } \mu \text{ bar)} - 88 \text{ dB (re 1 volt per } \mu \text{ bar)}$$

$$= -117 \text{ dB (re 1 volt)}$$

$$VL_{RCVR} = 1.4 \times 10^{-6} \text{ volts.}$$

In order to provide this voltage to the receiver at range 1 km, transmitter source level must be:

$$SL = SPL_{RCVR} + TL$$

$$SL = -29 \text{ dB (re 1} \mu \text{ bar)} + 75 \text{ dB}$$

$$SL = +46 \text{ dB (re 1} \mu \text{ bar).}$$

To realize this source level, the voltage level applied to the transmitter's transducer must be:

$$SL = \mathcal{SL} + VL_{XMTR}$$

$$VL_{XMTR} = 46 \text{ dB (re } 1\,\mu \text{ bar)} + [-36 \text{ dB (re } 1\,\mu \text{ bar per volt)}]$$

$$= 8 \text{ dB (re 1 volt)}$$

$$VL_{XMTR} = 2.8 \text{ volts RMS}$$

The transmitter output power necessary to provide this voltage level is:

$$VL_{XMTR}{}^2/|Z| = \frac{(2.8)^2}{|550 + j\ 10|} = .014 \text{ watts.}$$

In the final form of evaluation of the DUCS-1 transmitter, slightly over 50 volts (peak-peak) are developed across the transducer. Power output is:

$$P = \frac{\left(\frac{50}{2\sqrt{2}}\right)^2}{|550 + j\ 1C|} = \frac{(17.7)^2}{550.1} = .57 \text{ watts.}$$

This power level provides signal intensity to the receiver:

$$VL_{RCVR} = 9\,\mu \text{ volts.}$$

And: $\underline{\underline{S/N = 22 \text{ dB}}}$

It is re-emphasized that the above discussion is based on worst-case analysis. Maximum range, highest possible sea state for diving operations, and highest frequency components for most severe signal attenuation were the governing assumptions throughout the discussion. As can be seen, construction of the system as described should far surpass minimum design criterion.

V. SYSTEM DESCRIPTION

A. SYSTEM BLOCK DIAGRAM

The entire electronics package is housed in one underwater container and shares a common power source and a common transducer. In order to minimize the possibility of electromagnetic coupling and other undesirable feedback paths between transmitter and receiver, two completely separate modules have been constructed. One printed circuit board contains the entire transmitter, voltage regulator, and switching functions, while a second printed circuit board houses the receiver. The two boards are separated by a solid aluminum ground plane.

Figure 17 shows the system block diagram of the transmitter, receiver, housekeeping functions, and peripheral equipment. The use of circuitry common to the transmitter and receiver has been avoided wherever possible with the result that only the power source and switching functions are shared by transmitter and receiver.

The transmitter, built around a voltage controlled oscillator (VCO), applies to the transducer a 40 kHz frequency modulated voltage. The voltage developed in the microphone is amplified and applied as the control voltage to the VCO, which has linear voltage-frequency characteristics. The triangle output of the VCO is converted to a sinusoidal shape by a pseudo-logarithmic amplifier and is further amplified by the final amplifier. Provision is made for adjusting

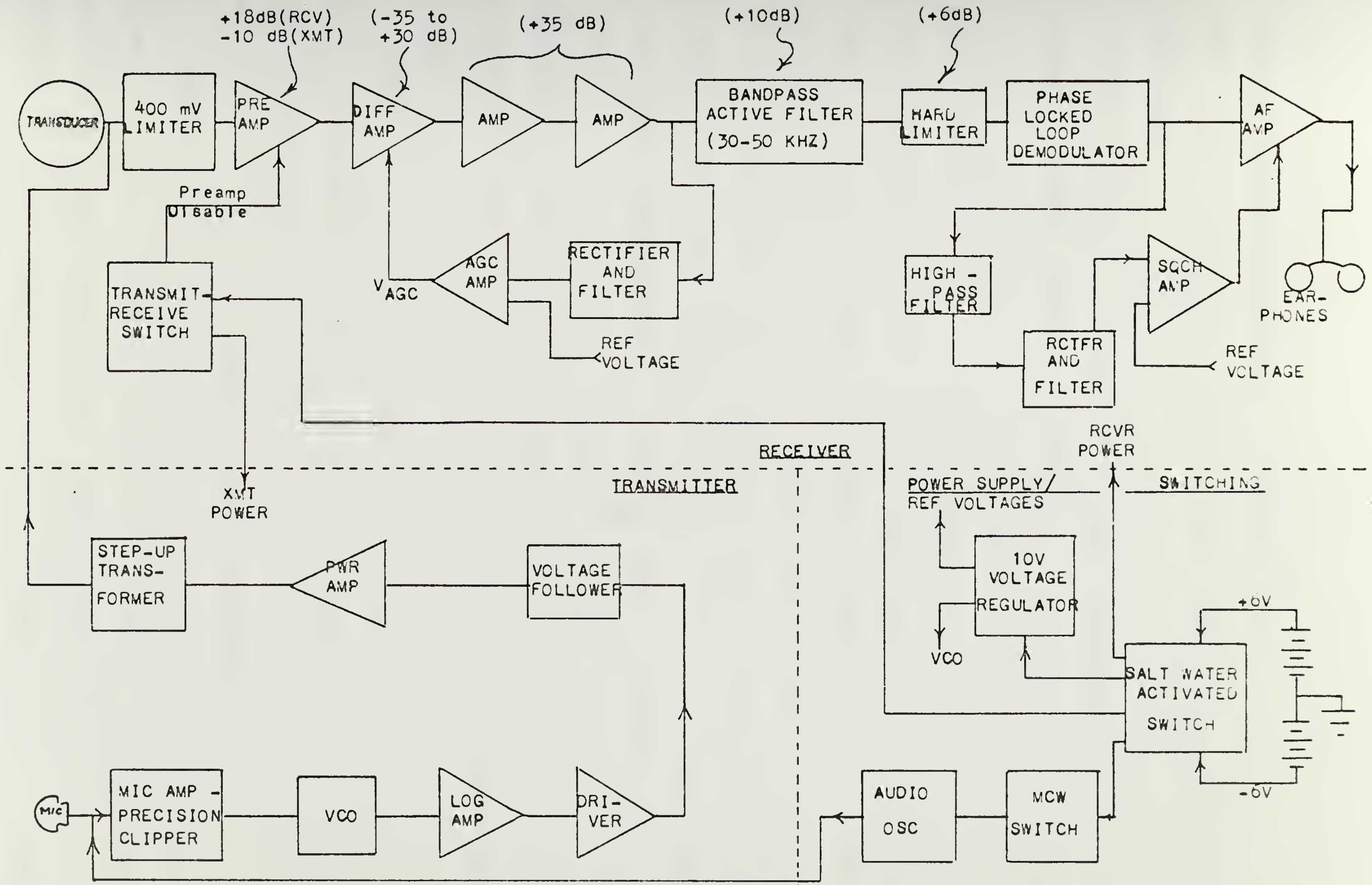


FIGURE 1// SYSTEM BLOCK DIAGRAM

frequency deviation from 0 to 10 kHz for test purposes. Additionally, microphone gain and VCO center frequency are adjustable.

The receiver is based on a single integrated circuit demodulator, a phase locked loop which develops an audio frequency voltage directly dependent on the instantaneous frequency of the received signal. Signals outside the bandpass (30 - 50 kHz) are attenuated by filtering, and that portion of the received signal which falls within the desired spectrum (30 - 50 kHz) is greatly amplified and limited before application to the input of the demodulator. The recovered audio intelligence at the output of the phase locked loop is amplified and fed to bone conduction earphones on the diver's head. Automatic gain control with wide dynamic range and squelch circuitry are additional features. Adjustments include AGC voltage, phase locked loop center (rest) frequency, squelch threshold, and volume.

Solid state switching provides automatic turn-on when the unit is immersed in water. The diver operates a push-to-talk button for plain voice transmission or a modulated CW button when transmission of a single tone is desired.

Transmit-receive switching is accomplished in the following manner: power is always applied to the entire receiver whenever the system is turned on. When transmitting, power is applied to the transmitter and simultaneously a bias voltage is applied to the receiver's first stage which causes that stage to attenuate the transmitted signal to such a level that

the remainder of the receiver can process it in the same manner as any received signal. This allows the diver to monitor his own signal as it is transmitted. It is thought that by a process of bio-feedback the diver may be able to improve intelligibility of his own outgoing signal.

B. DESCRIPTION OF CIRCUITRY

Table VII lists solid state devices referenced in schematic diagrams below.

1. Detailed Transmitter Analysis

a. Block Diagram

The transmitter block diagram is included as part of the system block diagram, Figure 17.

b. Voltage Controlled Oscillator

The voltage controlled oscillator (VCO) has been used as the heart of the transmitter circuitry. The model chosen was the Signetics NE/SE - 566, a single chip integrated circuit whose oscillator frequency is linearly dependent on applied voltage. Several checks of the linearity of this voltage-frequency relationship were made and excellent linearity was measured. See Figure 18.

C4 provides high frequency roll-off of the audio voltage that is in accordance with desired specifications. See Figure 8.

The rather elaborate voltage divider arrangement of resistors R11, R12, R20, and R10 provides some flexibility in center frequency adjustment. At the same time, it is

TABLE VII

Listing of Solid State Devices

I Transistors		II. Diodes		III. Integrated Circuits			
					Type	Company	Use
Q1	2N3706	D1 - D10	1N277	U1	UA 747	Fairchild	dual opamp*
Q2	2N3706	ZD1	1N961 (10.6V)	U2	CA 3019	RCA	diode array
Q3	2N3704	ZD2	1N750 (4.7V)	U3	SE 566	Signetics	VCO
Q4	2N404A	ZD3	1N750 (4.7V)	U4	UA 747	Fairchild	dual opamp*
Q5	2N3704			U5	CA 3028B	RCA	diff amp
Q6	2N404A			U6	UA 747	Fairchild	dual opamp*
Q7	2N3417			U7	UA 747	Fairchild	dual opamp*
Q8	2N3704			U8	UA 747	Fairchild	dual opamp*
Q9	2N3706			U9	SE 565	Segnetics	pll
Q10	2N5321			U10	UA 747	Fairchild	dual opamp*
Q11	2N3702						
Q12	2N5323						
Q13	3N187						
UJQ	2N4870						

*operational amplifier

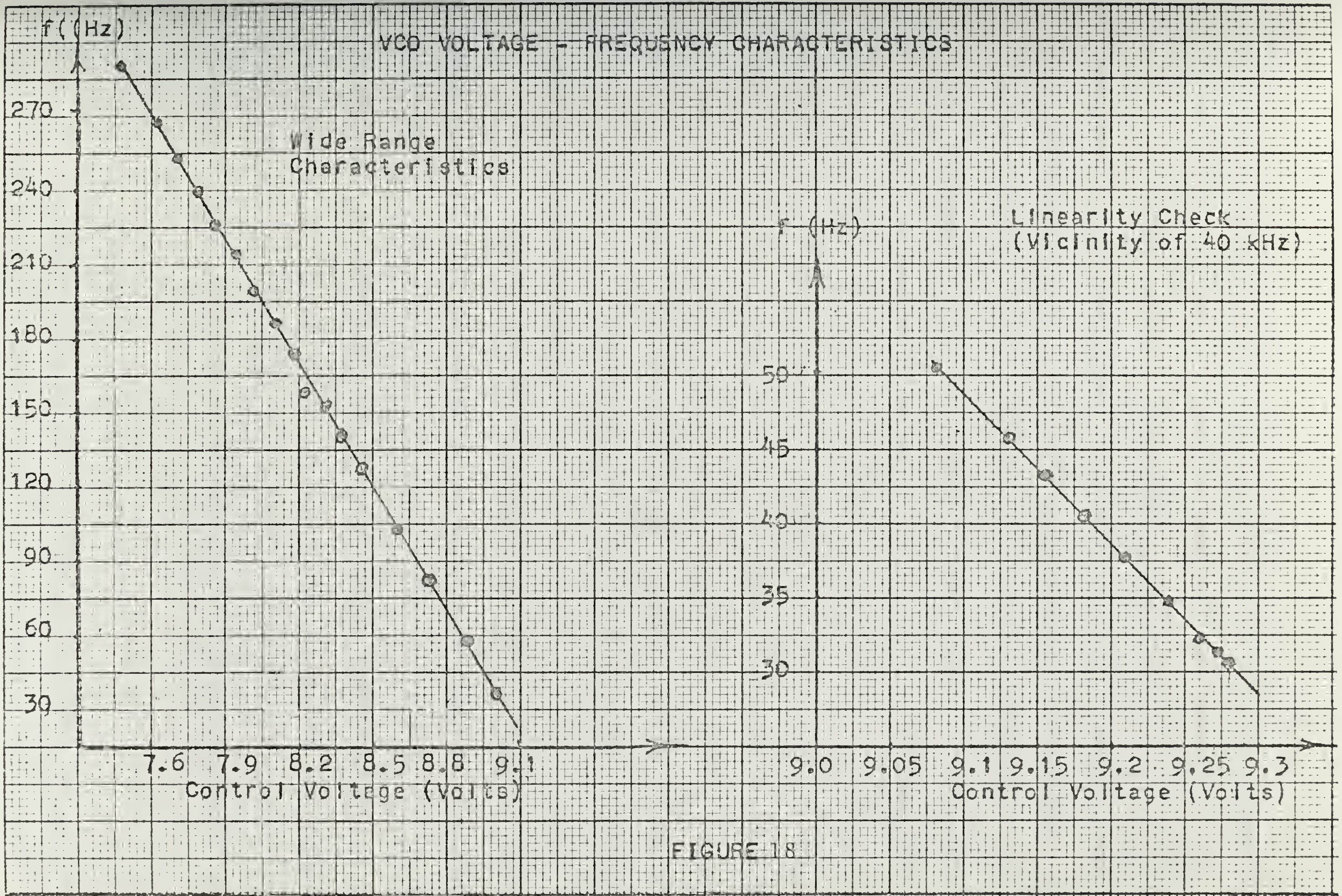

VCO VOLTAGE - FREQUENCY CHARACTERISTICS
f (Hz)
270
240
210
180
150
120
90
60
30
Wide Range
Characteristics
7.6 7.9 8.2 8.5 8.8 9.1
Control Voltage (Volts)
f (Hz)
Linearity Check
(Vicinity of 40 kHz)
50
45
40
35
30
9.0 9.05 9.1 9.15 9.2 9.25 9.3
Control Voltage (Volts)

FIGURE 18

impossible to mis-adjust the frequency to such an extent that it is completely out of the passband of the receiver.

The battery type chosen was nickle-cadmium penlight sized cells. A package of ten cells is quite compact while still providing better than 0.5 ampere-hours of energy. Measurements of the battery pack indicate that the usable range of supply voltage is 14.8 to slightly over 11 volts. Discharge is so rapid at voltages below 11 volts that even though the transmitter may continue to operate marginally, it will not continue to do so very long. Without voltage regulation, the center frequency of the VCO varied from 75.1 kHz to 24.5 kHz as the battery voltage decreased through its usable range. After the simple voltage regulator circuit was installed, the center frequency varied from 40.2 to 39.8 kHz over the same range of battery voltages.

Simple Fourier analysis indicates that a triangle wave such as that provided as an output voltage of the VCO is very rich in fundamental frequency content (87% f_o). It was therefore decided that since a sinusoidal waveform was desired, this output waveform would be utilized, allowing the square wave output to work into an infinite load for frequency measurements. The schematic diagram of the VCO is included in Figure 19.

c. Triangle-to-Sine Converter

Conversion to the triangle-to-sinusoidally-shaped carrier is accomplished by means of a single operational amplifier with non-linear feedback set up as a pseudo-logarithmic amplifier. Figure 19 shows the schematic diagram

VOLTAGE CONTROLLED OSCILLATOR
and
PSEUDO LOGARITHMIC AMPLIFIER

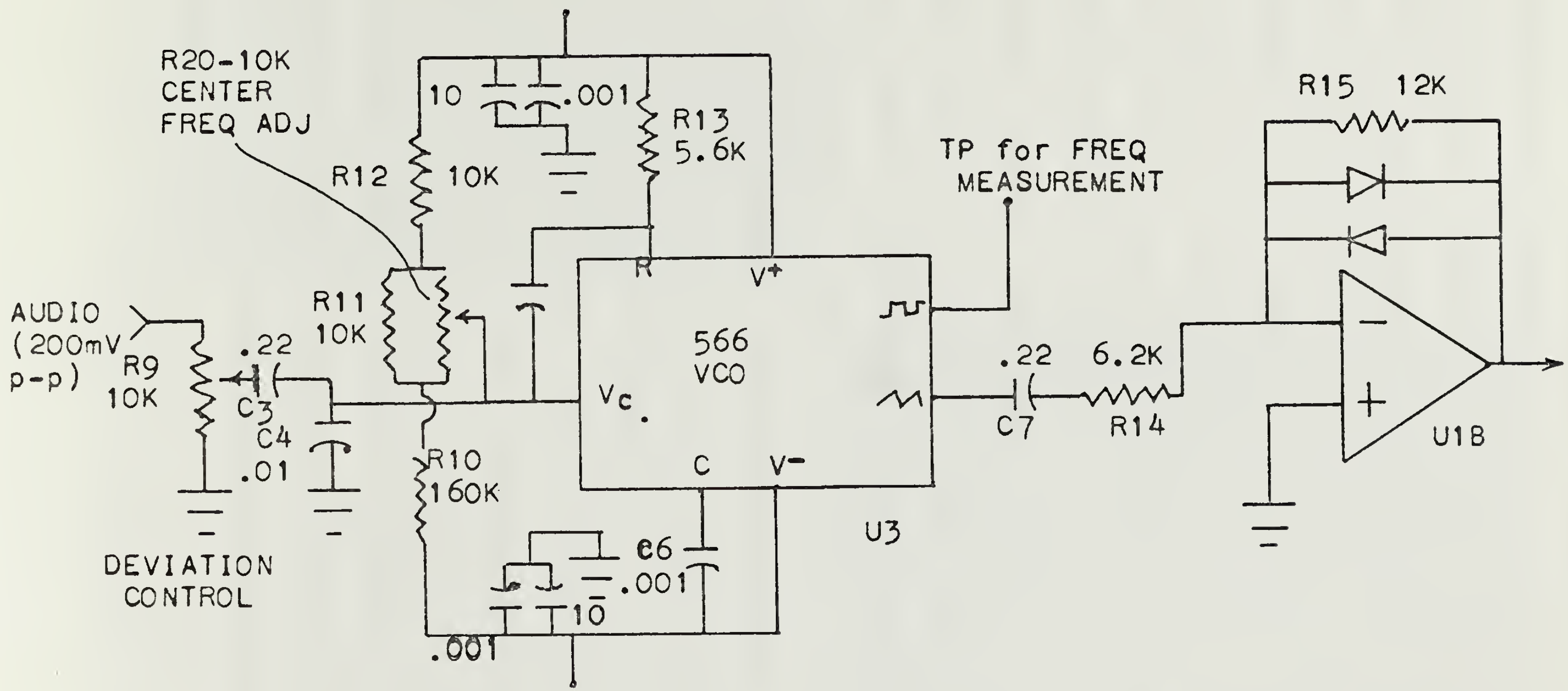


FIGURE 19

of this stage of the transmitter. This scheme was chosen over simple low-pass filter arrangements because of its ability to work into a rather low impedance final amplifier stage without appreciably altering the shape of the time description at the waveform. It was experimentably determined that loads as low as 30Ω could be accommodated by this scheme.

Figure 20 shows the near-sinusoidal shape of the waveforms as processed by this circuit. It is an actual photograph of an oscilloscope presentation of its waveform.

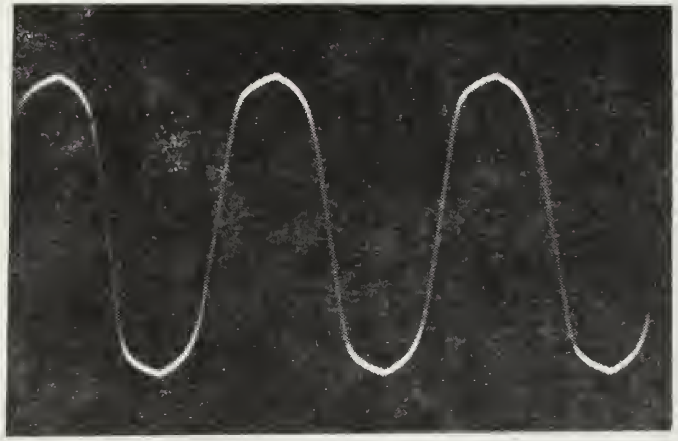
d. Final Amplifier Section

Figure 21 shows the schematic diagram of the final amplifier stage which consists of a voltage amplifier, a voltage follower, and a complementary pair of darlington-arranged pass transistors. The ratio R17/R16 sets up a voltage gain of approximately 19 dB which provides 3.2 volts (RMS) to the pass elements which deliver power directly from the batteries to the output transformer.

The voltage follower provides greater than 400 MΩ input impedance and only a few ohms output impedance. Included in its feedback loop are the pass-transistors Q_9, Q_{10}, Q_{11}, and Q_{12}. This scheme reduces the non-uniformity of the transistors by the multiplicative inverse of the amplifier gain.

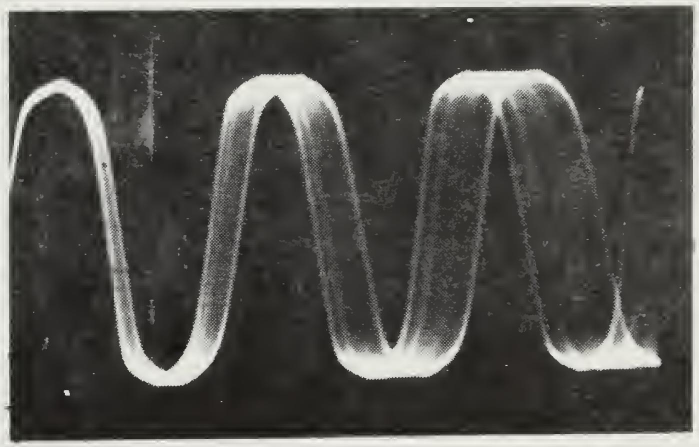
Darlington pass transistors were resorted to because of increased current gain and lower output impedance than that which could be realized by single power transistors.

TRANSMITTER WAVE FORM



40 kHz Carrier

No Modulation



40 kHz Carrier

1 kHz Tone

10 kHz Deviation

FIGURE 20

TRANSMITTER FINAL AMPLIFIER

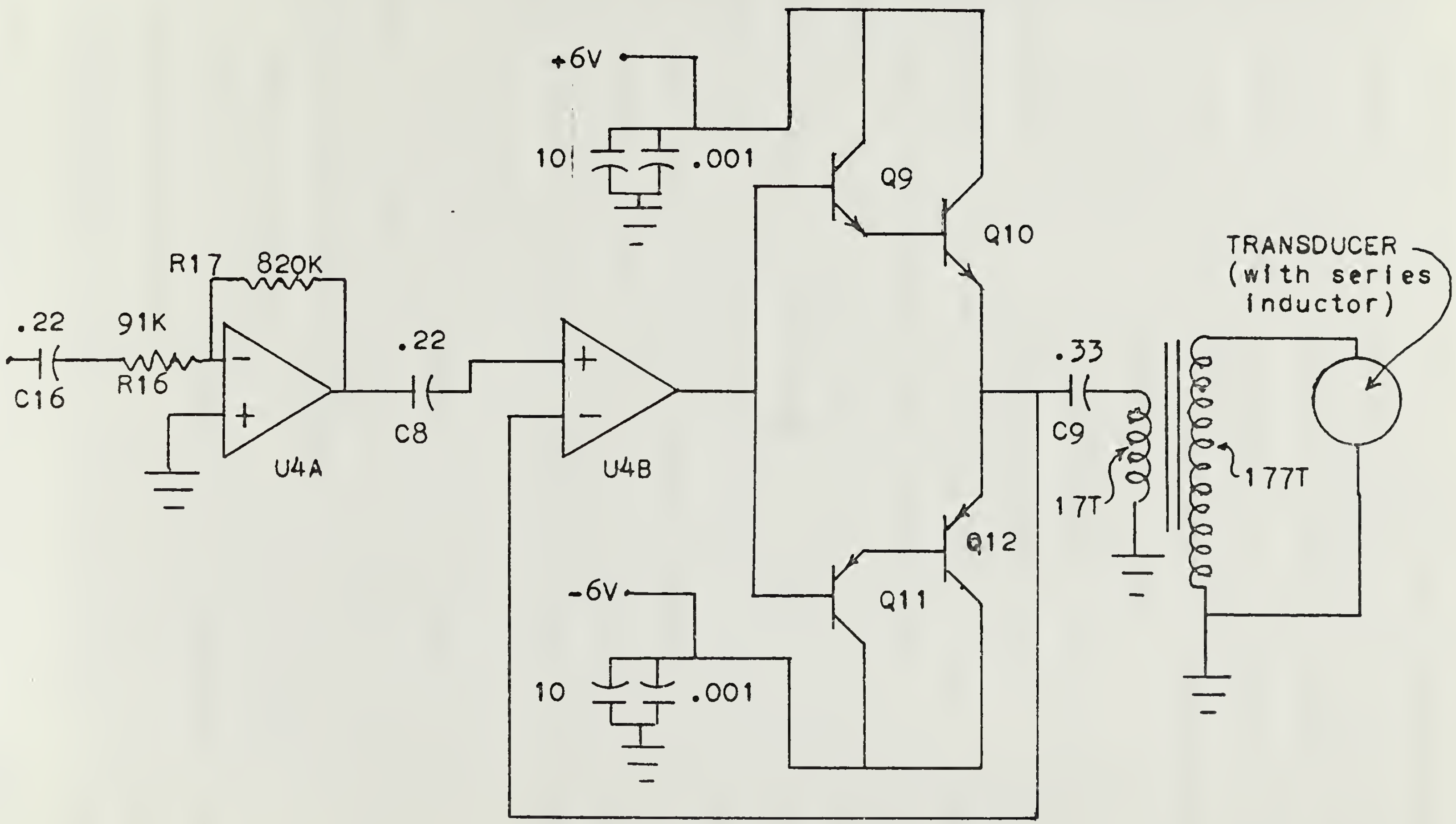


Figure 21

e. Microphone Amplifier, Precision Limiter

The precision limiter published by Burr-Brown[16] has been utilized as the microphone amplifier. This arrangement has transfer characteristics as shown in Figure 22. The schematic diagram is shown in Figure 23. It should be noted that R_L, as shown in Figure 11, includes the deviation control and the parallel input resistance of the VCO.

It is recognized that harmonic distortion of audio could be caused by hard clipping provided by this circuit. R_6 should be adjusted such that limiting never occurs except for unusually large peaks such as loud bubble noise or voice peaks when shouting. By utilizing this circuit, over-deviation of the carrier frequency is impossible. R_3 and R_7 were chosen so as to provide a maximum of 200 millivolts peak-to-peak, the audio signal applied to the control terminal of the VCO. As shown in Figure 6, this allows maximum deviation of the transmitted signal of $\pm$ 10 kHz. Figure 24 shows the frequency response of this stage.

Potentiometer R_9 allows the maximum deviation to be decreased to any value between 10 kHz and 0. This arrangement is schematically shown in Figure 7.

f. Audio Oscillator

Proponents of continuous wave transmission often point to its major advantage in support of their arguments; because of the remarkable signal processing possible in the

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[16]Graeme, J. D., Tobey, G. E., Huelsman, L. P., Operational Amplifiers, Design and Applications, p. 248, McGraw-Hill, 1971.

MICROPHONE AMPLIFIER
(Precision Bridge Limiter)

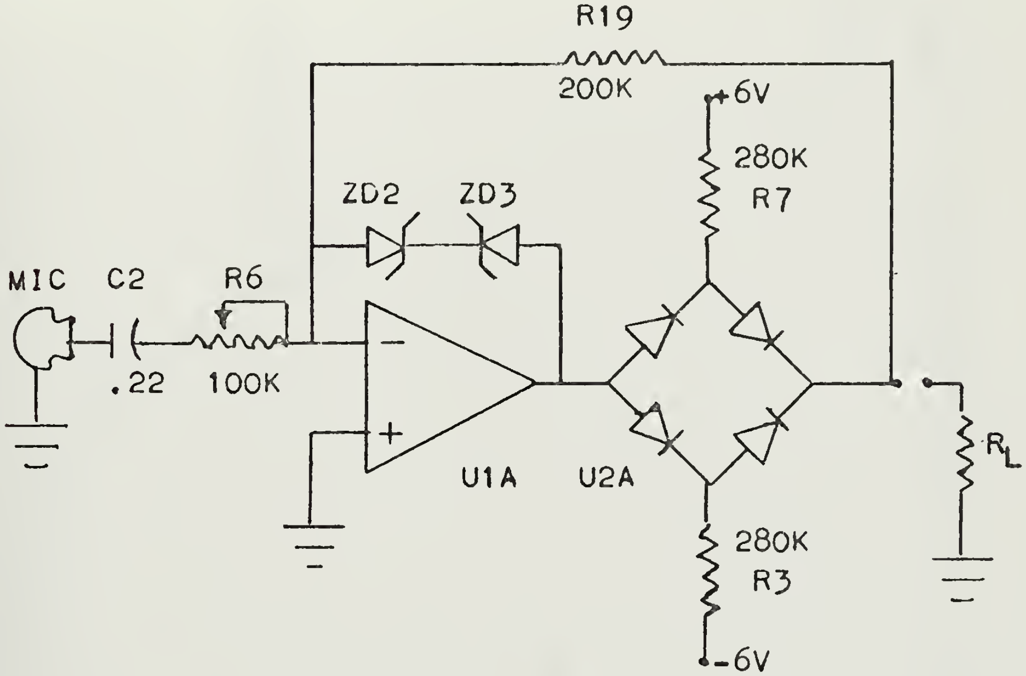


FIGURE 23

INPUT - OUTPUT CHARACTERISTICS
MICROPHONE AMPLIFIER

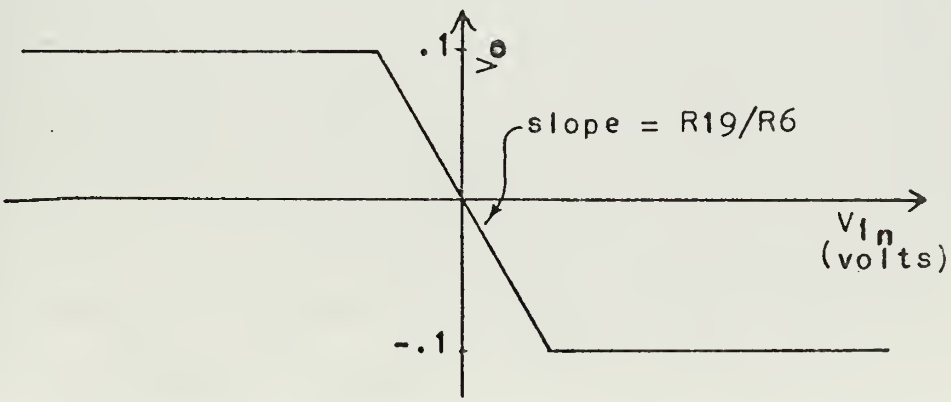
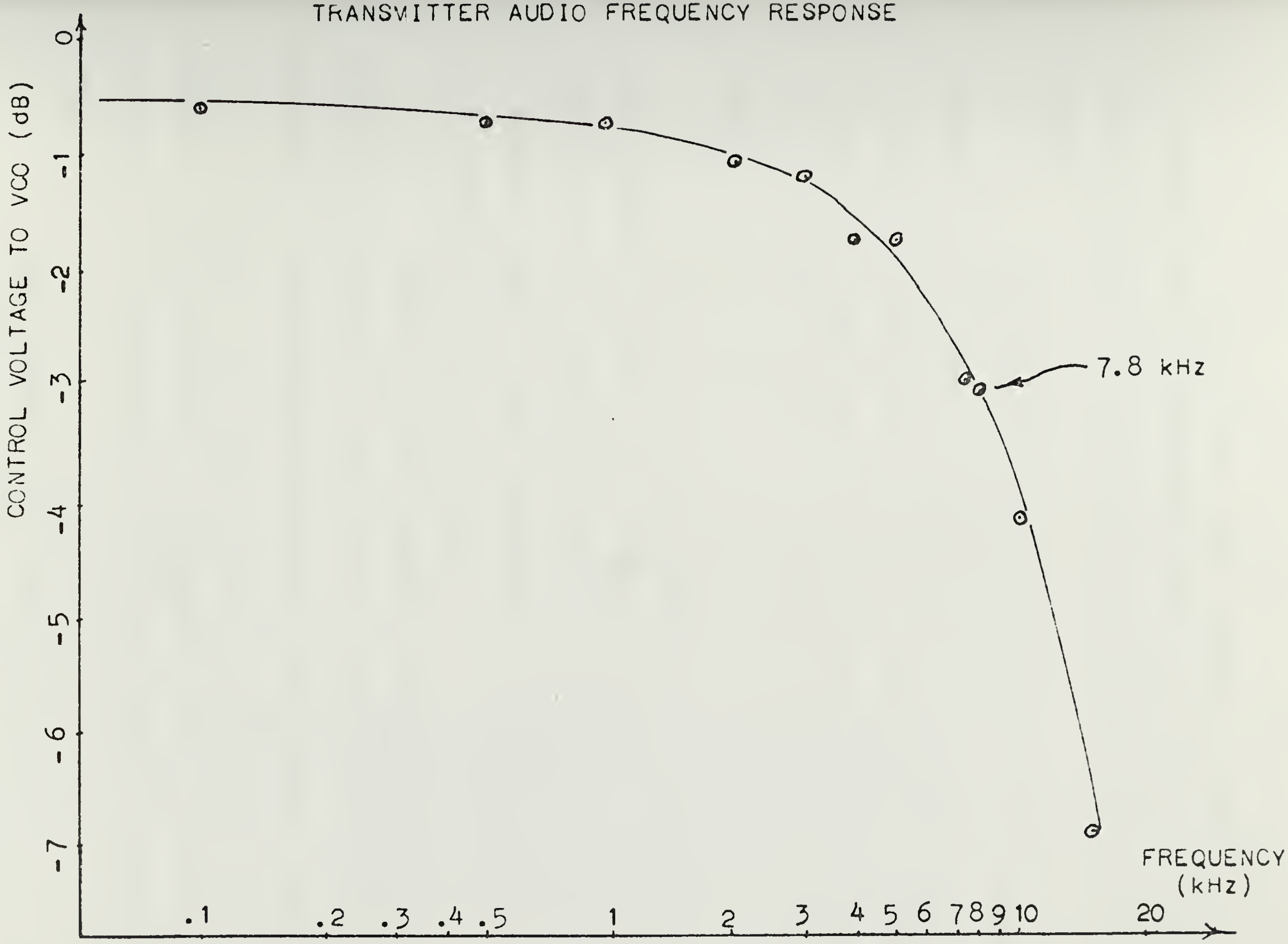


FIGURE 22

FIGURE 24
TRANSMITTER AUDIO FREQUENCY RESPONSE



receiver operator's nervous system, a single tone can be extracted from exceedingly small signal to noise ratios. It is thought that in the unpredictably noisy underwater medium, situations might occur which call for some mode of CW transmission.

Also conversations with active Navy salvage divers indicate that a simple surface-to-diver exchange such as "If you understand me, answer with two long tones ..." would be a significant improvement over presently available systems. It was therefore decided to include modulated CW in this project.

The uni-junction transistor is a four-layer (PNPN) semi-conductor device which has the ability to "fire" or switch rapidly from low to high conduction periodically. This device is ideally suited to the generation of distinctive audio tones of a relatively stable frequency.

Figure 25 is a schematic diagram of the uni-junction oscillator. Figure 26 is a photo of an oscilloscope picture of the waveform generated by the MCW oscillator. The time constant $\tau = R_2 \times C_{100}$, determines the approximate frequency of the oscillator. A frequency of 1.2 kHz was chosen because of its proximity to the most sensitive frequency to human ears.

The sawtooth wave developed across C_{100} is applied to the input of the microphone amplifier stage where it is processed as any other audio signal. In fact, while using MCW mode the microphone is not disconnected and the diver can intersperse tones with plain speech if he desires to do so.

PUSH - TO - TALK AND MODULATED CW

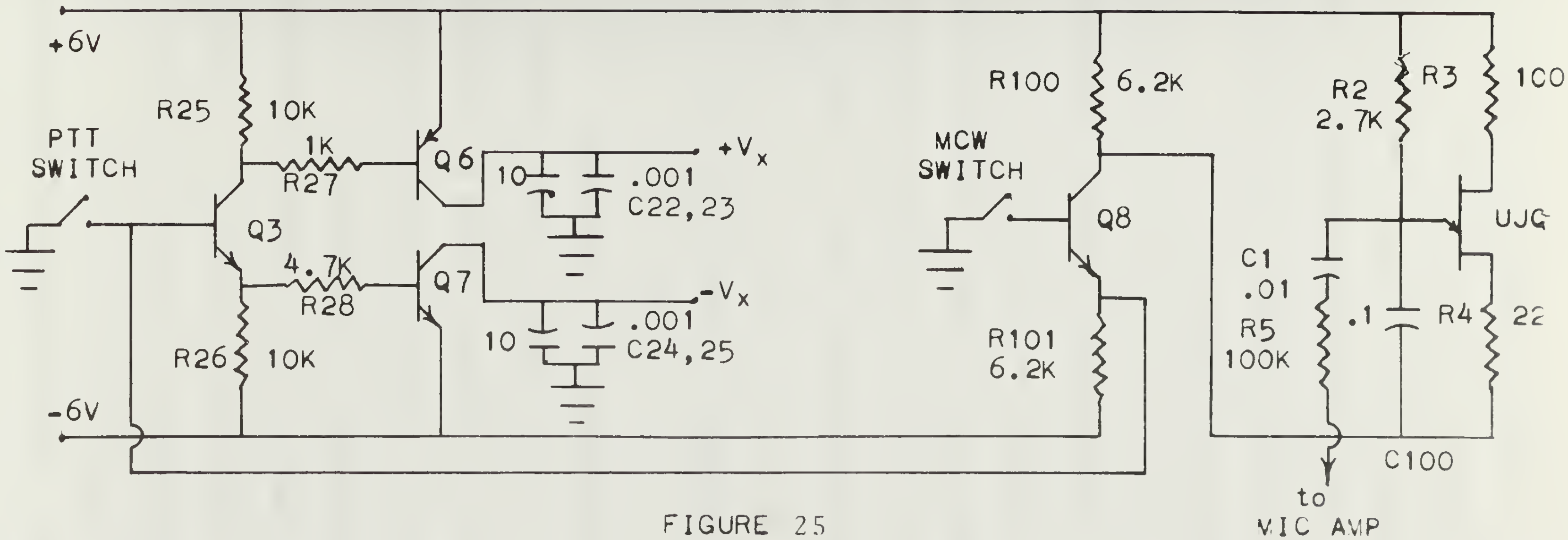
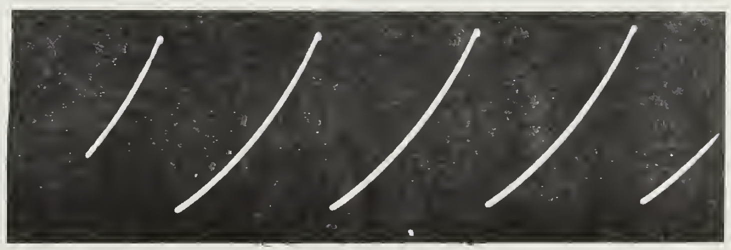


FIGURE 25

MODULATED CW WAVEFORM

FREQUENCY - 1.25 kHz



AMPLITUDE - 0.6 V_{p-p}

FIGURE 26

g. Transformer Design

As discussed in detail above, the transducer impedance was experimentally determined to be $550 + j\,10\,\Omega$. Since the output impedance of the power amplifier is quite low, transformer coupling was indicated.

The transformer was wound on an AMIDON T80-2 torroid form. The primary consists of 17 turns of #18 wire, the secondary is 177 turns of #23 wire. Turns ratio is therefore 10.4. This ratio was chosen to present a load impedance of approximately 5 ohms to the power output stage of the transmitter, causing it to draw slightly over 1 watt from the battery pack. The voltage level across the transducer is roughly 50 volts (peak-peak).

2. Detailed Receiver Analysis

a. Block Diagram

Because of its simplicity, low signal-to-noise requirements for good demodulation, and its overall superior performance, a phase locked loop has been used as the demodulator for the DUCS receiver. Since good selectivity characteristics can be provided by the demodulator itself, little would be gained by the use of the usually necessary superheterodyne techniques. Therefore, the entire receiver, except for the audio section, operates in the 40 kHz band.

Since incoming signal RMS voltages in the range 9 microvolts to 70 millivolts are expected, and a design criterion of voltage level provided to the phase-locked loop is 0.35 volts (RMS) total maximum gain from transducer to demodulator

must be in the order of 88 dB with AGC dynamic range of 78 dB. Q_{13} provides about 18 dB and U_5 provides from -50 to +30 dB of gain depending on AGC voltage applied. U_6 provides about 35 dB and U_8 provides about 6 dB.

Figure 17 shows the block diagram of the receiver and its inter-relationship with the entire communication system.

Bandpass characteristics are provided by the transducer itself, which operates near resonance, and by careful selection of coupling and bypass capacitors. Also, use has been made of an active bandpass filter which severely attenuates frequencies outside the passband. Because of the nature of absorption losses in sea water, the expected received spectrum could have as much as 12 dB difference in amplitude between the lowest and highest frequency components. Because of this and other noise limiting desired, a hard limiter stage was added just ahead of the phase-locked loop.

b. Pre-Amplifier Stage

In order to utilize the transmit-receive switching scheme explained below, the first stage of the receiver must necessarily have high input impedance. The low noise figure and very low current drain characteristics of insulated gate field effect transistor technology dictate the use of a device of this type. The use of a dual gate device provides additional transmit-receive flexibility; by changing the voltage of gate number 2, the device can be biased into pinch-off, thus providing about 10 dB attenuation, and allowing the remainder of the receiver to function normally while

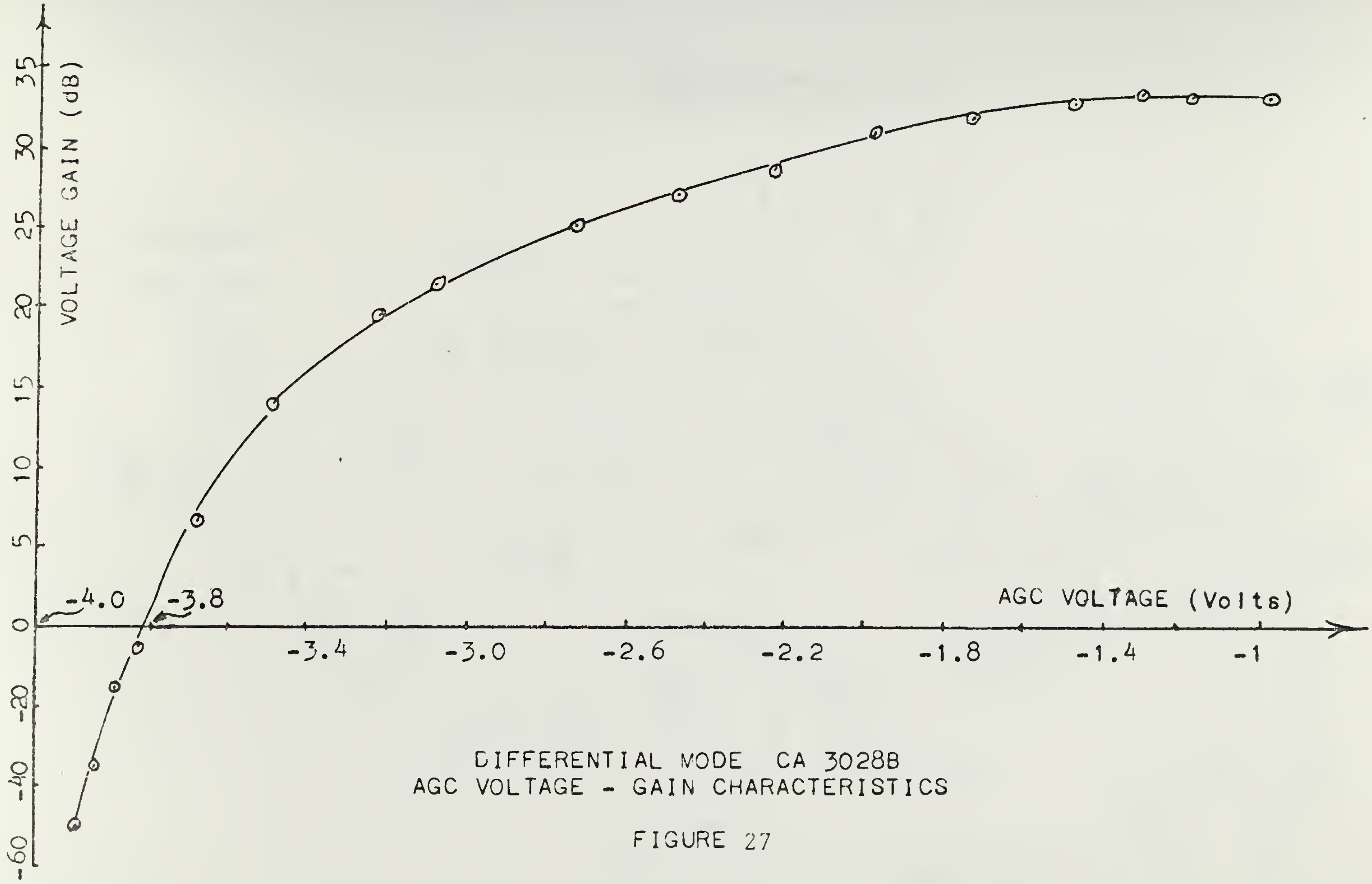
transmitting, thus allowing the diver to monitor his own transmission as it is sent to the transducer. This scheme has the additional advantage that a continuous check of the entire receiver is accomplished each time the diver transmits.

The second amplifier stage of the "front-end" section of the receiver utilizes the popular three transistor array: CA3028B. The circuit is arranged as a differential amplifier to provide additional gain after the MOSFET first stage. A significant advantage of using this integrated circuit is its wide dynamic range; measured gain variation is in excess of 80 dB. By simply varying the voltage applied to the resistive voltage divider network of the constant current source transistor, voltage gain varies as shown in Figure 27.

Figure 28 shows a schematic diagram of the receiver "front-end." Since both devices used in this section are useful at much higher frequencies than needed in this application, careful attention has been paid to reducing the high frequency response of both sections. The final design utilizes a double-sided circuit board with one side un-etched. This side acts as an effective ground plane and hence unwanted feedback paths in the high gain first stages are greatly attenuated. As a result of these precautions, the receiver did not oscillate.

c. Automatic Gain Control

Figure 29 is the schematic diagram of the AGC rectifier, filter, and amplifier. D_3 and D_4 and C_{23} provide a standard rectifier/voltage doubler. The combination of



DIFFERENTIAL MODE CA 3028B
AGC VOLTAGE - GAIN CHARACTERISTICS

FIGURE 27

RECEIVER FRONT END

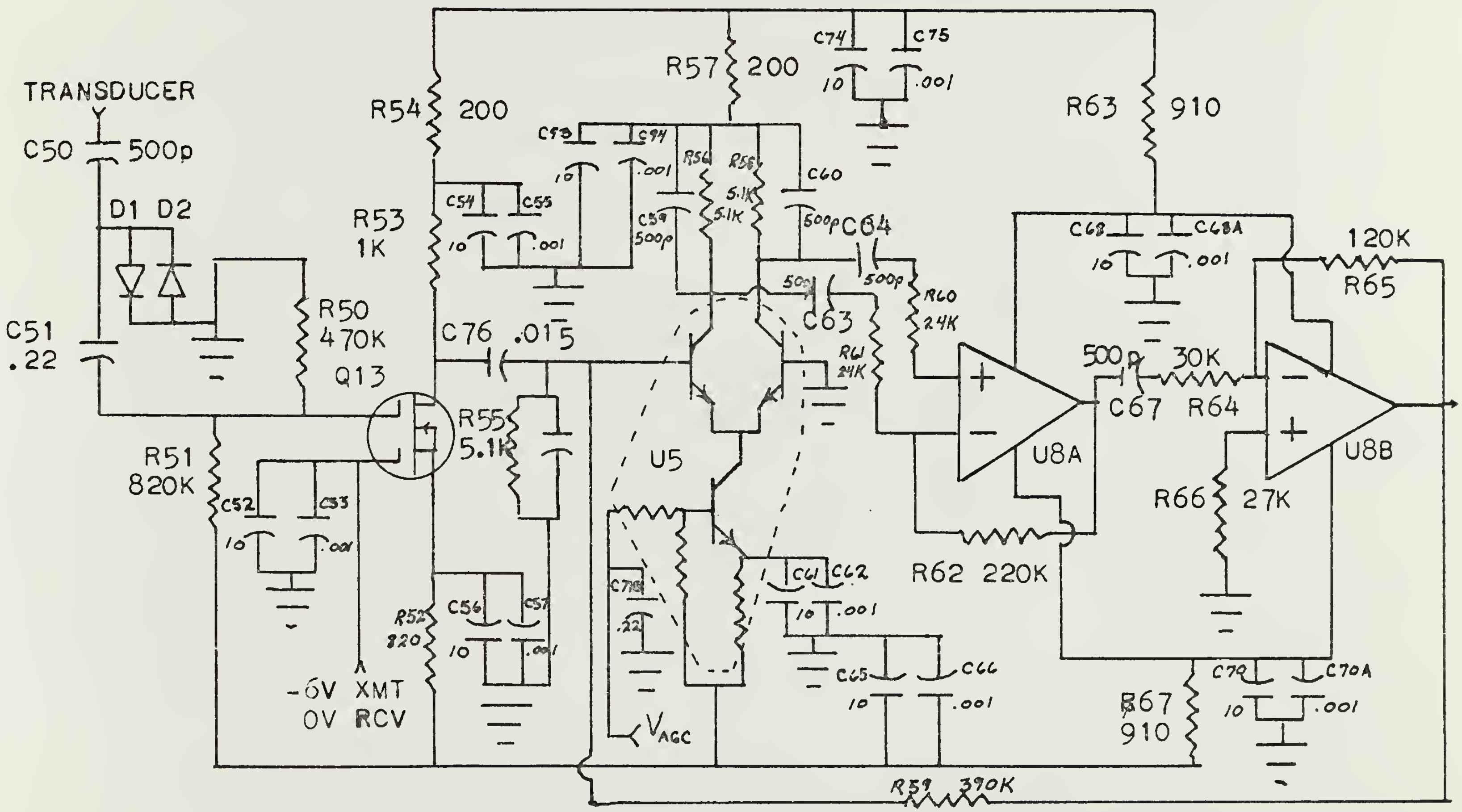


FIGURE 28

AUTOMATIC GAIN CONTROL AMPLIFIER

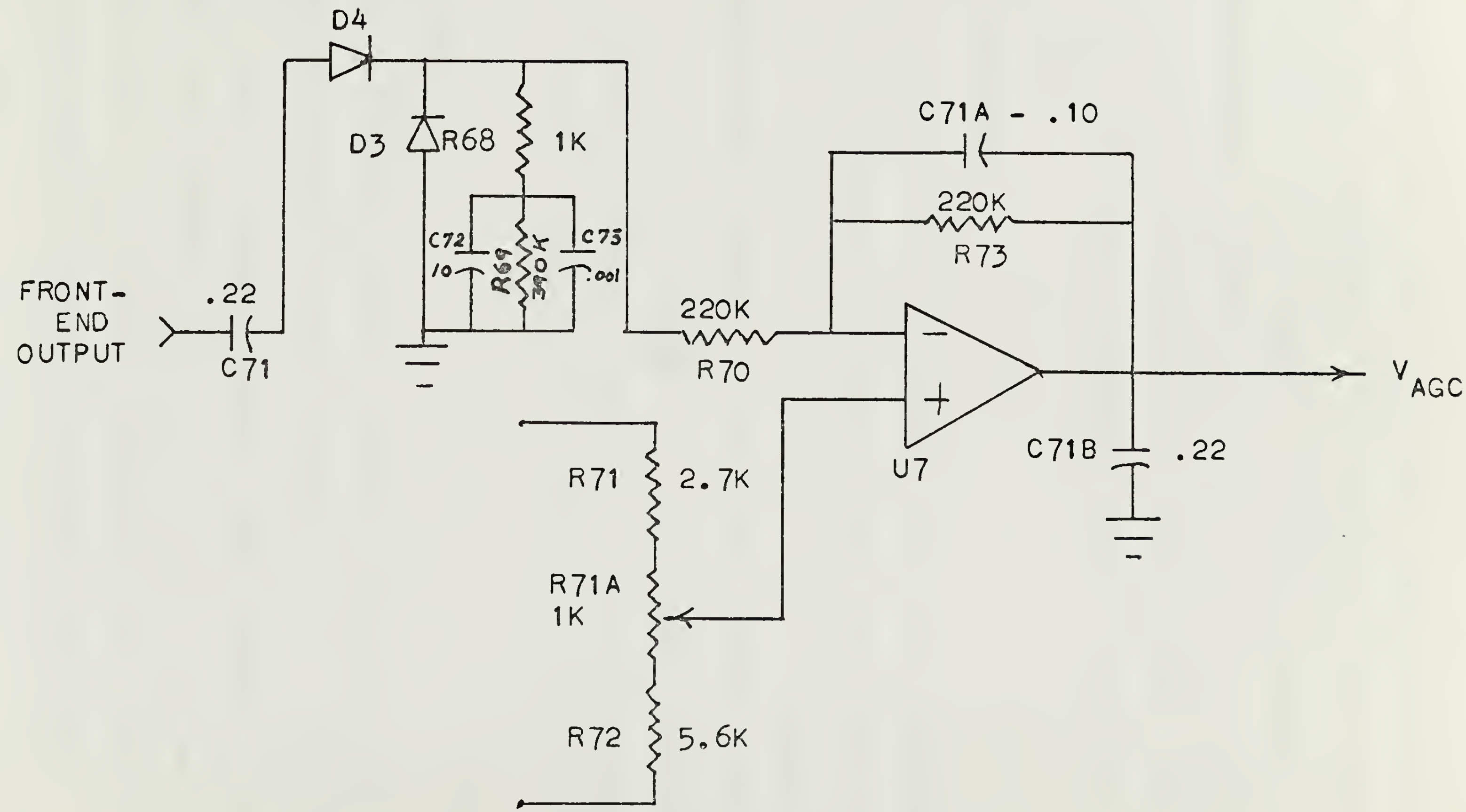


FIGURE 29

R_{32}, C_{35} and R_{31} provides good AGC timing characteristics and U_7, a 741 operational amplifier, is set up as a DC integrating amplifier to provide AGC voltages in the desired range for the CA3028B. The reference voltage is a fraction of the regulated voltage provided from the transmitter circuit board. This scheme allows for relatively unchanged operation over the range of battery voltage.

d. Active Filter

A standard active bandpass filter was designed such that its 3 dB bandwidth is about 20 kHz with center frequency of 40 kHz. A 12 dB per octave roll-off is realized. This stage additionally provides about 10 dB voltage gain. Figure 30 is the schematic diagram for this stage and Figure 31 shows its measured frequency response. The filter makes use of 1/2 of a 747 dual operational amplifier integrated circuit.

e. Hard Limiter

The precision limiter circuit utilized as the microphone amplifier/limiter and described in the transmitter section of this paper is again used as a hard limiter in the receiver. This stage provides absolute maximum amplitude of 0.5 volt (peak-peak) to the phase locked loop, and provides about 6 dB voltage gain. Figure 32 shows the schematic diagram of the stage.

f. Phase Locked Loop Demodulator

The extraction of modulating voltage from a frequency modulated carrier has taken many forms since Armstrong

ACTIVE FILTER

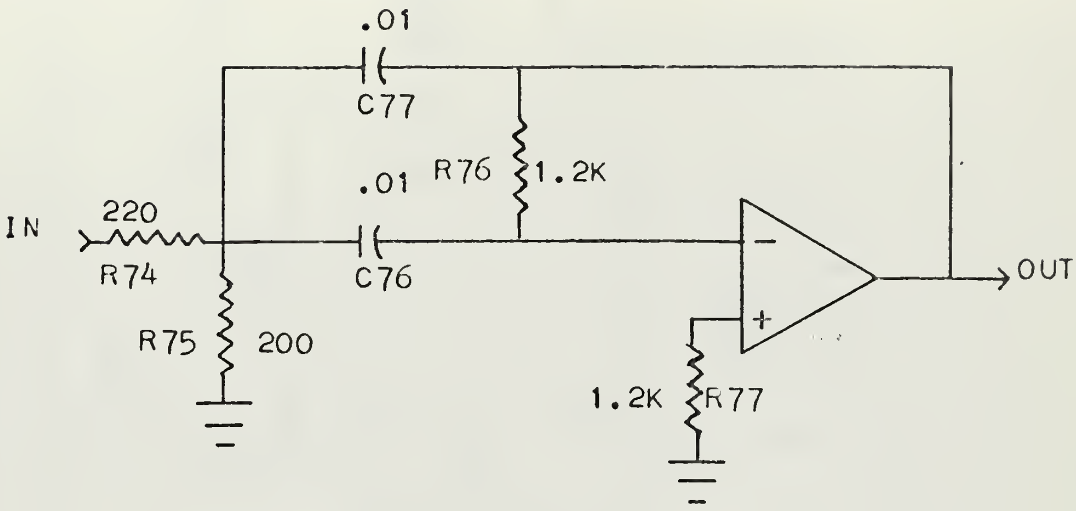


FIGURE 30

ACTIVE FILTER FREQUENCY RESPONSE

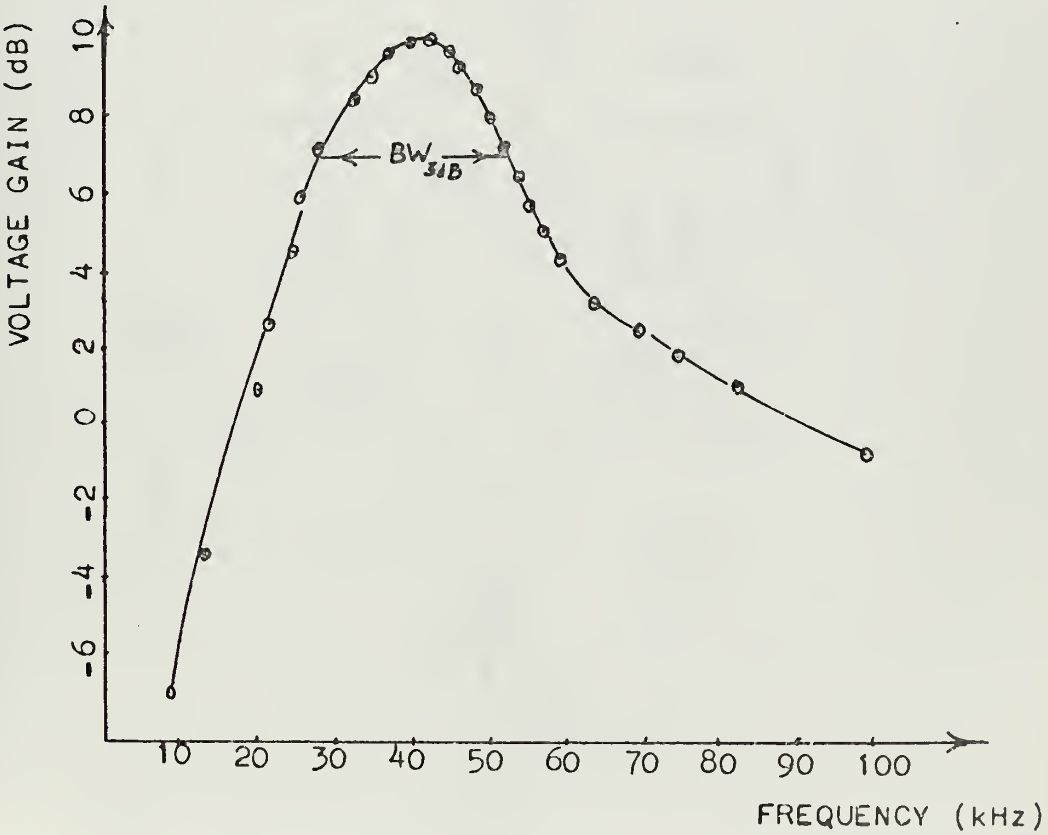


FIGURE 31

HARD LIMITER AND PHASE LOCKED LOOP DEMODULATOR

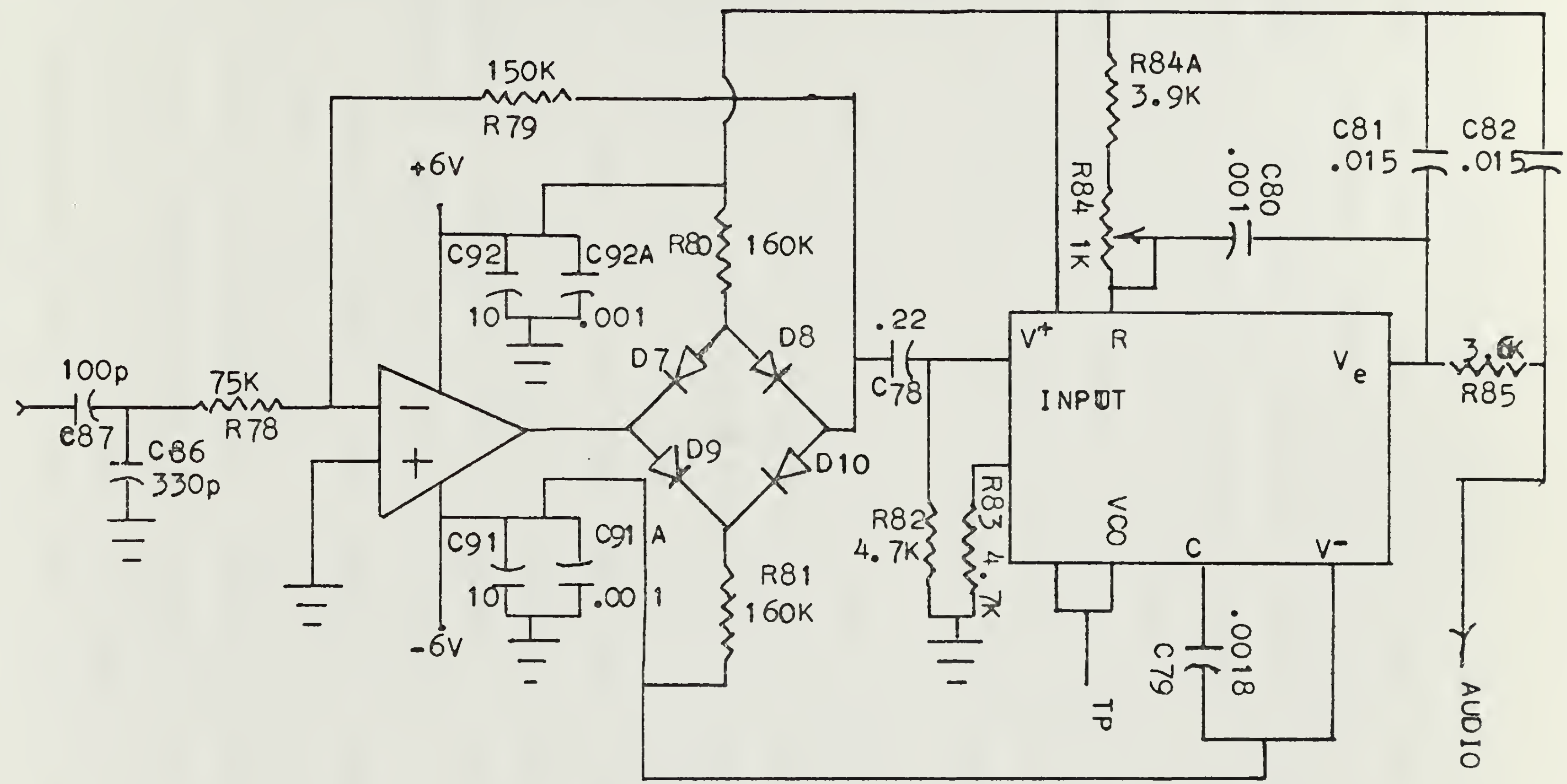


FIGURE 32

spearheaded the advance of FM technology. One of the techniques which works very well is the use of the phase locked loop which is based on frequency feedback technology dating back perhaps forty years. This type of demodulator provides ease of adjustment of center frequency and bandwidth, excellent selectivity, and, most significantly, superior performance in high noise environment. Early applications of phase locked loops were implemented in discrete component form and, even with the advent of the transistor, such circuitry was rather complex and bulky.

Recently monolithic integrated circuits have been developed which combine the functions of phase comparison, limited filtering, and voltage-controlled oscillator in a single miniature package. The PLL requires no resonant tuned circuits, or inductors of any sort, which makes it well suited to applications in small, highly portable equipment. The model chosen for use in the DUCS-I receiver is the SE/NE-565, manufactured by Signetics, Sunnyvale, California.

Numerous books have been written to exhaustively analyze the phase locked loop and since no original additions to this literature are offered here, this discussion will be brief. The basic block diagram of the phase-locked loop is shown in Figure 33. As can be seen, the phase-locked loop is basically an electronic servo loop which, by generation of an error signal attempts to drive its internal oscillator to the exact phase of the input voltage. Assuming a perfectly linear voltage-frequency characteristic of its voltage-controlled

PHASE LOCKED LOOP BLOCK DIAGRAM

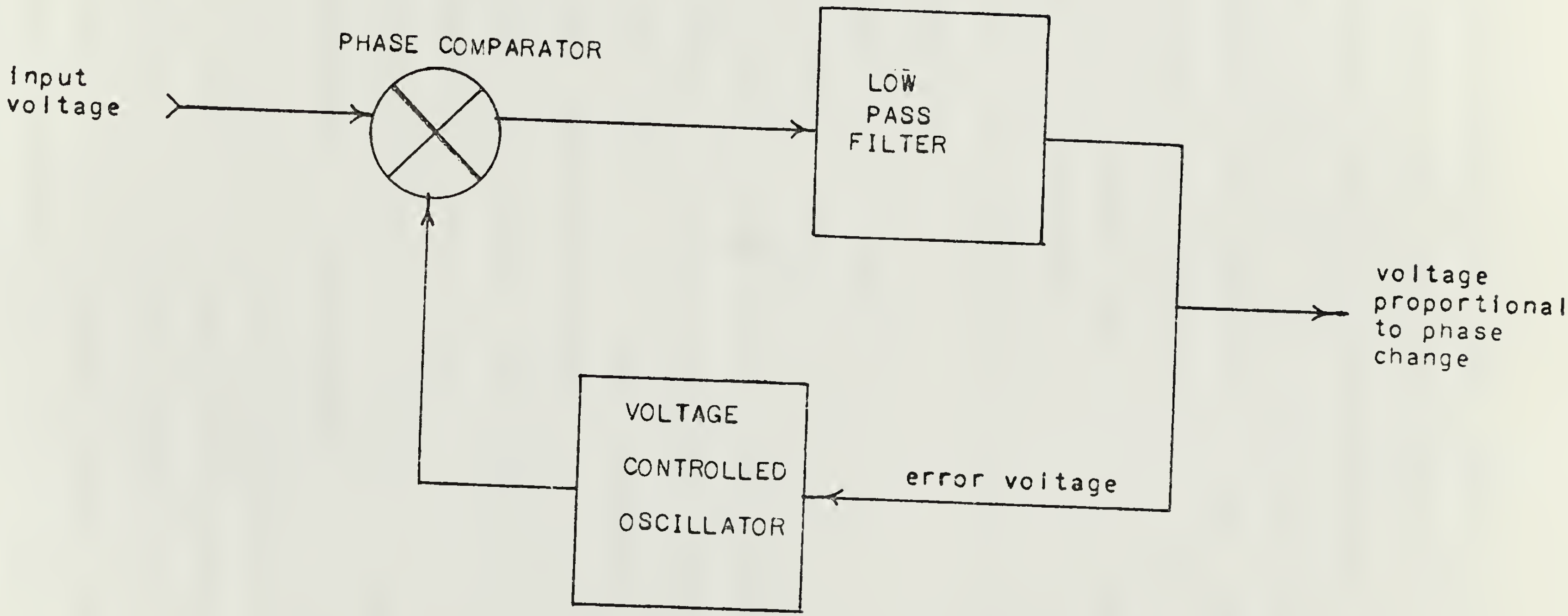


FIGURE 33

oscillator, the error voltage is an exact replica of the modulating signal for the input frequency modulated carrier.

The schematic diagram of the demodulator circuitry is included as part of Figure 32. R84, R84A, and C79 determine the rest frequency of the demodulator's VCO. Adjustment of R84 allows minor changes in rest frequency. With component values as shown, measured capture range is 28 to 63 kHz. Lock range was experimentally determined to be 22.5 to 70.4 kHz. No visible distortion of the demodulated waveform was noticed in oscilloscope presentation.

g. Audio Amplifier and Squelch

The demodulated voltage, the message audio, requires amplification before application to the earphones. Figure 34 is the schematic diagram of the audio amplifier and squelch circuit. Unfortunately, the bone conduction earphones used by the system have a measured impedance of only 3Ω, so transformer coupling to the earphones was indicated. A small audio output transformer fills this role.

Audio from the demodulator is forced through the high pass filter combination of R87A and C85. It is then applied to the rectifier/voltage doubler arrangement of C84, D5 and D6 and finally filtered by C86. The DC potential applied to the inverting input of U10B is thus proportional to the amplitude of hiss (lack of quieting) found in the audio. Because of the very high ratio R89/R88, U10B acts as a switch providing power to the audio amplifier when quieting occurs (only when a signal is received) and effectively disabling the

AUDIO AND SQUELCH CIRCUITRY

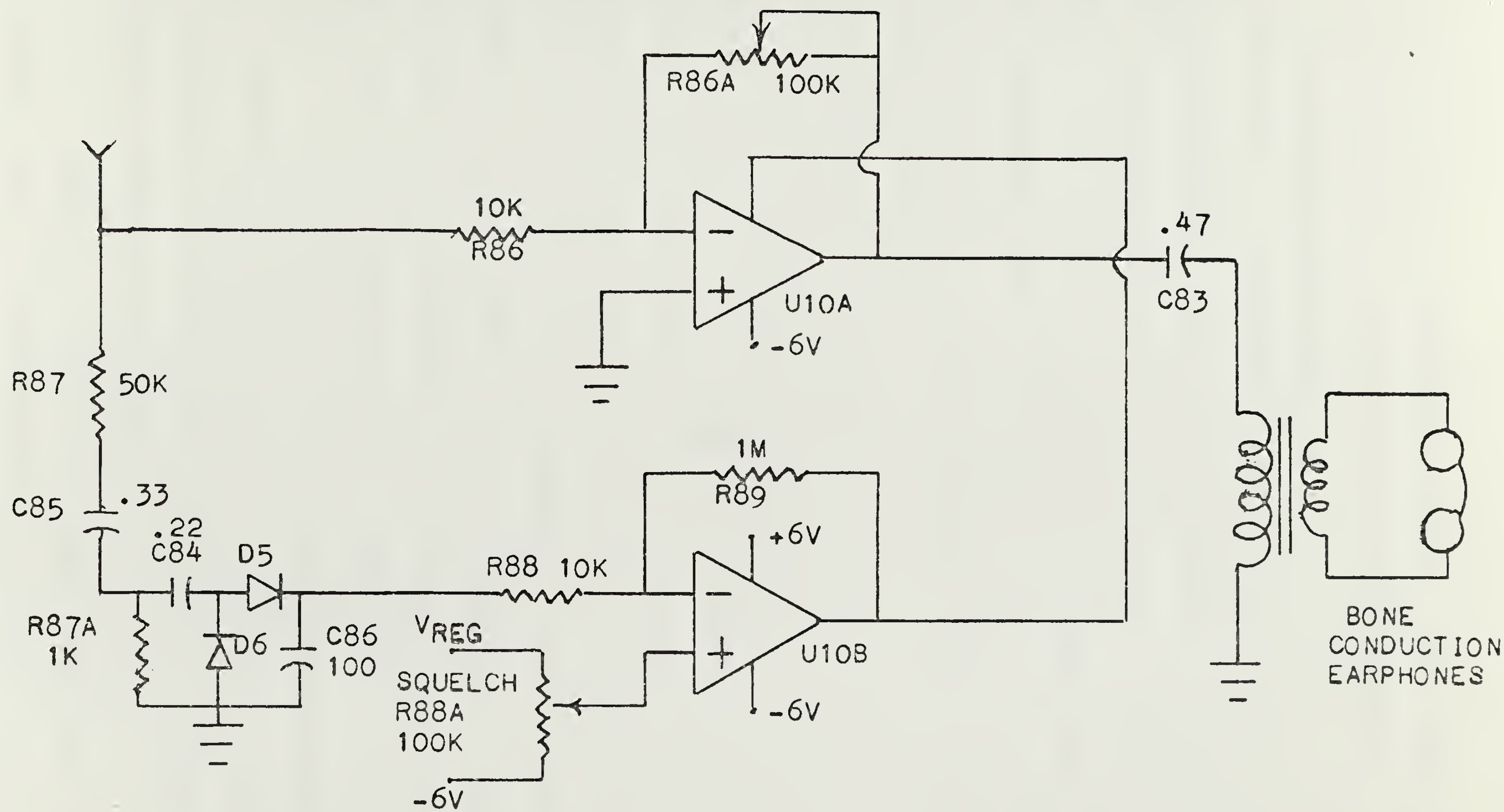


FIGURE 34

audio amplifier when no quieting occurs. Very sharp squelch cut-in characteristics are realized in this manner.

3. Housekeeping Functions

Included on the transmitter circuit board is the ten volt voltage regulator, salt water activated on-off switching circuitry, modulated CW switching circuitry, and push to talk switching. These stages are classified as performing "housekeeping" functions. The system block diagram (Figure 17) exhibits the inter-relationship of these stages to the working of the entire transceiver.

a. Solid State Switching

(1) Salt-Water Activated Switch. In order to preclude the possibility of inadvertent complete battery discharge after a diving operation as well as to avoid the necessity of breaking water tight integrity of the housing for a power switch, automatic off-on switching is an original feature of the communication system. A brass conductor is installed in the end cap of the underwater housing and is electrically insulated from the metallic housing. When the unit is immersed in sea water, ionic conduction of the medium places the brass fitting at ground potential. Figure 35 shows this arrangement. Precise machining of the parts was necessary to prevent possible water leakage at pressure.

Figure 36 shows the solid state switching scheme used for the salt-water activated power switch. Transistor Q_1 is normally non-conducting because the base and emitter are both at a potential of -6V; hence, since no

AUTOMATIC ON/OFF
SOLID STATE SWITCHING

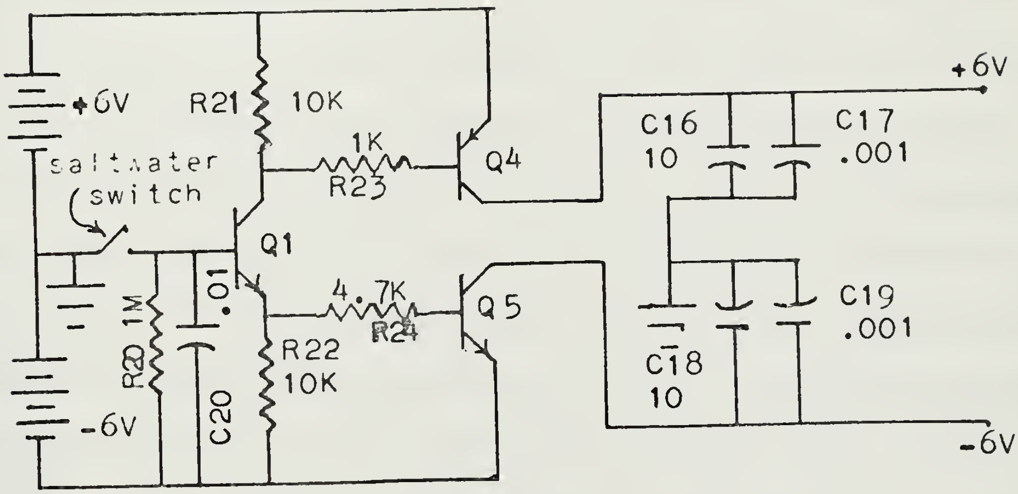
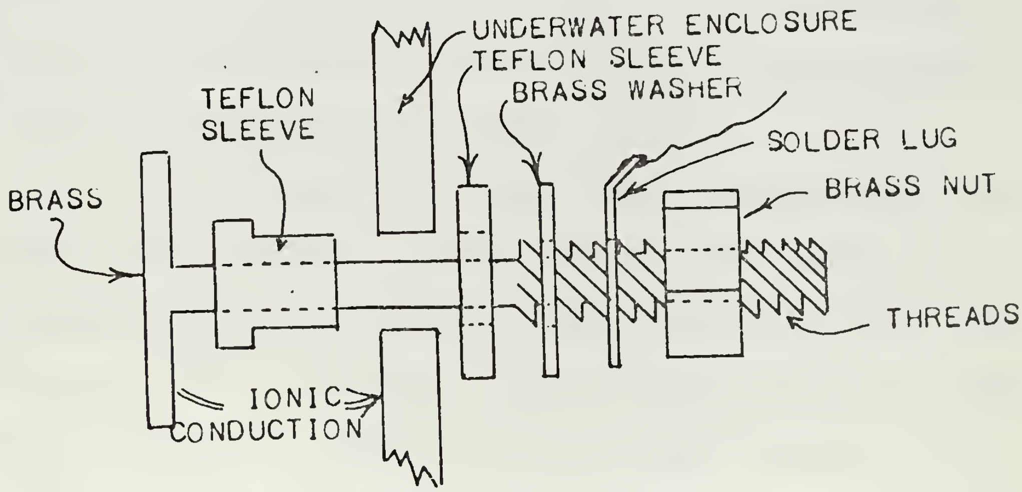


FIGURE 36

FIGURE 35

SALT-WATER SWITCH



collector current is flowing in Q_1, the collector and emitter voltages are +6V and -6V respectively. When the base is put at ground potential (OV), the base emitter junction of Q_1 is effectively forward-biased driving the collector and emitter voltages toward ground potential. This likewise saturates both Q_4 and Q_5 by placing a forward bias on their base-emitter junctions. Q_4 and Q_5 were chosen by investigating a large sample of transistor types on the curve-tracer to obtain the devices which required the least collector-base reverse bias for operation. That is, transistors were chosen that exhibited the smallest voltage drop between collector and emitter when in the saturated mode (less than 0.2V in each case). This makes almost the entire battery voltage available to transmitter and receiver circuitry when the set is turned on.

(2) Push-to-Talk. The same scheme as described for the salt water activated switch is used for the push to talk switching function. Instead of salt water ionic conduction, the base of transistor Q_3 is forced to ground potential by a hand operated underwater switch which is large and easily operated even when wearing bulky rubber gloves.

The transmit-receive scheme utilized by this communications system never disables the receiver which operates continuously whenever the salt water activated switching circuit is in the "on" condition. The +6 and -6 volts at the output of Q_6 and Q_7 (see overall circuit block diagram, Figure 17) are applied to the transmitter and only the first stage preamplifier section of the receiver.

(3) <u>Modulated CW Switching.</u> In addition to the push to talk switch, the diver has the option of closing a second switch manually. This switch, the modulated CW switch, performs a dual function; not only does it enable the previously described uni-junction audio oscillator circuit, but also it grounds the base of Q_3, thus enabling the push to talk circuitry. This makes it unnecessary to push two buttons, one creating the tone and one to enable the transmitter. Figure 26 illustrates the MCW switching circuitry.

When the switch is closed, the transistor goes into saturation. This brings both the collector and emitter voltages near ground potential and activates both the push to talk switching circuit and the audio oscillator uni-junction circuit.

b. Voltage Regulator

In order to minimize dependence of transmitter frequency, and reference voltages, on the battery voltage, it was found that a voltage regulator would be necessary. A voltage near 10 volts was desired as that is the lowest reliable operating voltage of the VCO. A 10.6 volt zener diode was thus chosen as the reference followed by a somewhat high current gain transistor in common collector arrangement as a pass element. Figure 37 shows the schematic diagram of this stage.

The dropping resistor biases the zener diode with less than 1 milliamp of current; the zener diode has been measured on a curve tracer to ascertain that it will provide

VOLTAGE REGULATOR

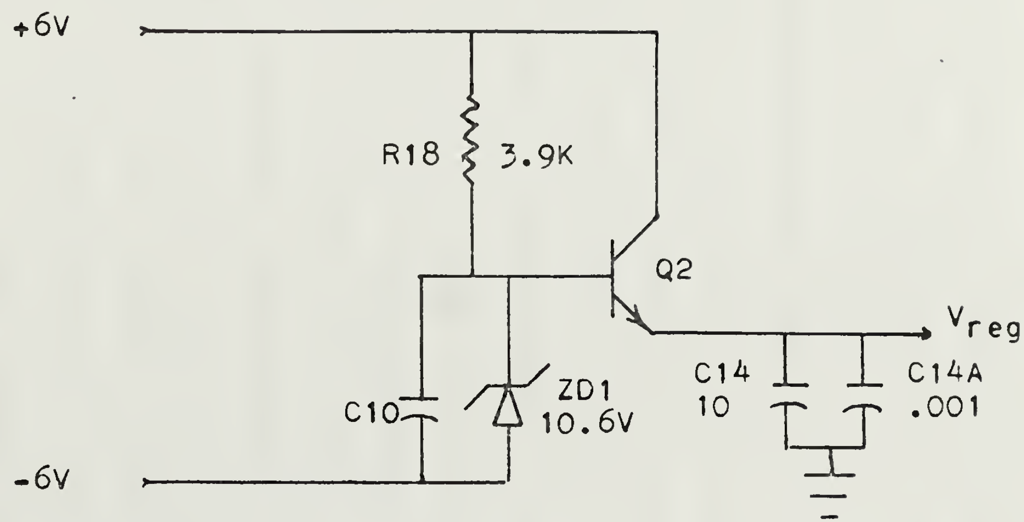


FIGURE 37

good regulation at such low current. The high gain ($h_{fe} \approx 400$) transistor allows rather significant changes in the load impedance and/or changes in supply battery voltage to affect the voltage across the zener diode very little. In fact, it was found that by varying the battery voltage over its entire operating range (14.8V to about 11.8V), the changes in emitter voltage at Q_2 were not measurable.

c. Transmit-Receive Switching

Figure 28 indicates how the single transducer is used by both the transmitter and the receiver. The 500 pF capacitor provides roughly 8KΩ of reactance to current coming in from the hydrophone. Since neither of the diodes will conduct until signal levels near 400 mV (peak-peak) occur, these diodes provide a very high impedance to ground. Since, with the transmitter disabled, the step-up transformer primary "sees" very high impedance, the secondary of that transformer can be considered effectively infinitely high in impedance. Thus, virtually all of the incoming current flows toward the receiver and by voltage-divider logic, practically all of the voltage at the transducer appears across gate number 2 of the MOSFET preamp.

On transmit, a very large voltage is applied to the 500 pF capacitor which, by its large reactance, limits the current applied to the clipping diodes. Thus, around 400-500 mV (peak-peak) is actually applied to the preamp input. By virtue of its pinch-off mode, -10 dB attenuation is realized in this stage; around 44 mV (RMS) is available

at the drain (output) of the preamp. This is well within dynamic range of the AGC and hard limiter circuitry.

VI. SYSTEM EVALUATION

A. ENCLOSURE AND PERIPHERAL EQUIPMENT

Several dives have been conducted in the Monterey and Carmel bays using the underwater communication system described. The first three dives were made with empty enclosures to ensure watertight integrity before installing electronics. Maximum test depth has been sightly below 75 feet (22.9 meters). No leaks of any sort were noted.

Once in the water, the enclosures are slightly positively buoyant. Except for the tendency of the leg straps to ride down the smooth, nylong-covered wetsuit material on the leg, the mounting is comfortable and provides minimum encumbrance to the diver. However, out of the water this arrangement is quite awkward and uncomfortable. Mounting the enclosure directly on the diver's air tank is also possible simply by adjusting the straps to its diameter.

The underwater enclosure is constructed entirely of aluminum which has been painted with zinc chromate primer and a lacquer finishing coat. However, some corrosion is evident especially in the vicinity of bi-metallic junctions at the base of the underwater connectors and the transducer housing.

The full face mask used is uncomfortable with microphone installed. This is because this model of face mask does not have enough volume for proper installation of a microphone. If the mask were modified to incorporate the microphone into

its enclosure and if integral cable fittings were installed, the mask and microphone combination would be more acceptable. Noise cancellation and frequency response of the microphone are quite impressive.

The muzzle, used with a double hose regulator, performs fairly well but jaw fatigue is considerable. The aforementioned problems with speech cavity design and distortion are noticeable.

Earphones are uncomfortable, at best. Every time the diver turns his head, the earphones tend to twist under his wetsuit hood, making them ineffective much of the time. Also, the earphones tend to slip down from the temple area to the fleshy part of the neck. A better design for the earphones would include some means of holding them securely against the mastoid bone behind the ear.

B. TRANSMIT-RECEIVE EVALUATION

Battery life is in excess of five hours under key-down conditions. All electronic circuitry performed as designed with the exception of the squelch circuitry. There may be problems in the design of its trigger circuitry.

Two-way communications have been achieved at ranges in excess of one hundred meters. Audio quality is excellent in that users have commented that they can clearly recognize the voice of the transmitting diver. Although intelligibility is generally very good, further tests using the Modified Rhyme Test or some similar means of applying a meaningful quantitative evaluation to the system are required.

On several occasions the system has been tested in relatively quiet conditions over a sandy bottom and no problems were encountered in securing good signal to noise ratio. Further testing over different bottom compositions and in generally noisy conditions (surf and wind) should also be accomplished.

In short, the system performs well as designed and built. It is apparent that intelligibility and reliability improvement can best be accomplished by attention to the peripheral equipment.

VII. RECOMMENDATIONS FOR FUTURE IMPROVEMENT

A. ENCLOSURE AND PERIPHERAL EQUIPMENT

Although for many diving situations it may be more comfortable and convenient to mount the underwater enclosure on the diver's air tank than on his leg, this may lead to a degradation of communication. The transmission impedances of air and water are so widely separated that any body of air in the path of the outgoing acoustic wave in sea water will severely attenuate its propagation in that direction. The lungs and face mask both contain volumes of air and should be well removed from the radiation axis of the transducer. Leg mounting provides this separation: tank mounting does not.

Some problems with tank mounting are secondary. First, when swimming on the surface using snorkle, a standard activity in the course of almost any SCUBA dive, the transducer is often completely out of the water. Sufficient dampness can remain on the salt water activated switch to keep the unit power turned on. Damage could conceivably occur in the transmitter final transistors if the diver were to transmit with the transducer in air; the transducer impedance is far lower in air than in water. The second problem is particularly troublesome in the Monterey area. Heavy kelp growth often entangles the SCUBA diver and is very difficult to remove from behind the head. Perhaps the ideal method of attaching to the diver's body would be the incorporation of a large pocket sewn on the pant leg of his wetsuit, or affixed to his weight belt.

Bi-metallic junctions should be avoided in the construction of the communication system. As mentioned in the evaluation section above, even with very careful attention to rinsing in fresh water and towel drying, corrosion was hard to abate. Greater use of plastics bears investigation.

A cylindrical shape for the housing would have been far easier to manufacture and to make watertight. Forcing "O" rings to seal around square corners is far more difficult than the use of round "O" ring grooves.

The problems encountered with masks, microphones and earphones should be rectified early in any testing program. Perhaps the Kirby-Morgan KMB-8/10 mask which has already gained wide acceptance with Navy divers would prove to be more comfortable than the present arrangement. Cable splices should be avoided whenever possible.

The single most important method of improving the system appears to be the judicious choice of microphone, mask, and earphones. The basic design criteria of keeping the system low in cost and yet still acceptable to the user should be kept in mind when choosing this equipment. Earphones of a much higher impedance would eliminate the need for the small audio output transformer used.

B. ELECTRONICS DESIGN

The overall design and layout of the transmitter-receiver appears to make effective use of the small volume provided by the enclosure. The separation of the transmitter and receiver both electrically and physically has worked well.

Several changes in the design should be investigated. First, in order to improve S/N still further and at the same time increase maximum range, two changes could be incorporated. A larger battery pack could provide more output power. With the present underwater enclosure, this would require very little mechanical change; room exists for a few additional cells around the equipment. Also, instead of making efforts to provide the transducer with a sinusoidal voltage, a far more efficient arrangement would be to develop pulses in the transmitter and operate the final amplifier in a manner analogous to Class "C" operation. A one-shot multivibrator triggered by the square wave output of the VCO could easily accomplish the generation of pulses. The resonant nature of the transducer will severely attenuate harmonic content of the transmitted waveform and only the fundamental (40 kHz sinusoid) will be launched. This arrangement could eliminate the four output transistors by making use of a switching element such as a member of the silicon controlled rectifier (SCR) family. In this manner, higher power and low current drain could be realized with relative ease.

A modification to the system which would be easy to incorporate yet could provide significant intelligibility improvement would be the incorporation of pre-emphasis in the FM transmitter and de-emphasis in the receiver. Simple low pass and high pass circuitry could be added to accomplish this.

C. CONCLUSIONS

After a thorough study of current literature, and the experience of building and testing DUCS-I, the author is firmly convinced that the basic format of the system is the optimal approach to filling the need for underwater communications for divers. That is, transmitting a 40 kHz frequency modulated signal received by a high-gain receiver which utilizes a phase locked loop demodulator represents a conceptual and realized improvement over competitive systems.

A great deal of work in the design of good man-machine interface equipment is called for. Microphones and earphones that have low distortion and yet are comfortable to use would enhance the intelligibility of DUCS-I considerably.

Improvements in enclosure design and modifications to the circuitry as proposed above may somewhat improve its performance and acceptance by the user.

Much more thorough testing of the prototype units by actual fleet divers engaged in Navy diving operations would undoubtedly result in several additional recommendations for improvement. It is recommended that this testing be accomplished in the very near future.

It is further recommended that the engineering expertise of civilians and active duty Navy personnel currently engaged in electronics and underwater design be used to the fullest extent possible to bring the system to a production model.

APPENDIX A: SYSTEM PHOTOGRAPHS

Figures 38 through 45 are reproductions of Polaroid photographs of the actual system, including all peripheral equipment. Printed circuit artwork is included in case the reader may desire to construct a similar system for testing and evaluation.

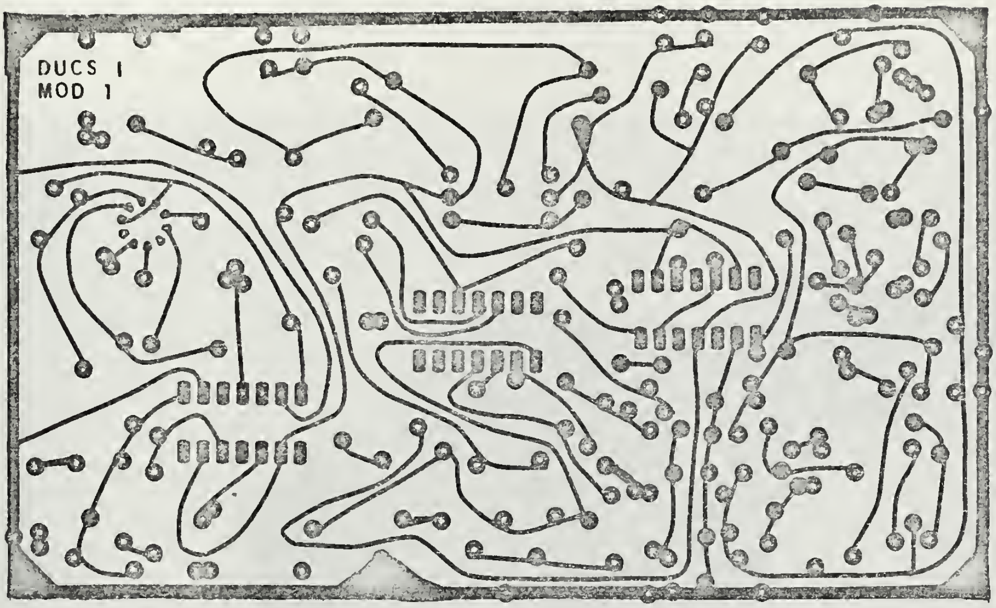
FULL FACE MASK

FIGURE 38

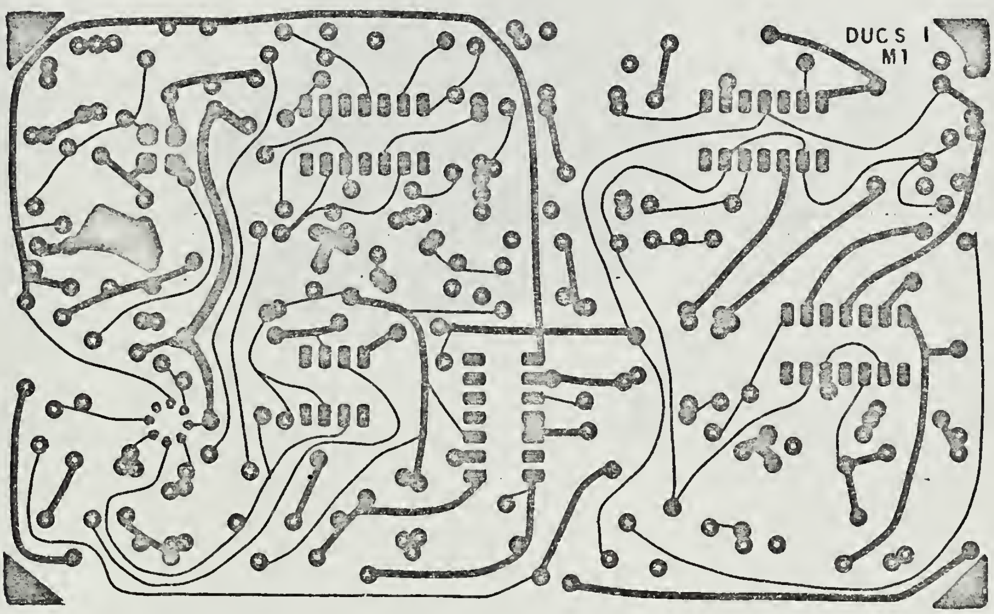
MUZZLE AND DOUBLE HOSE REGULATOR

FIGURE 39

PRINTED CIRCUIT BOARD LAYOUT



TRANSMITTER



RECEIVER

FIGURE 40

THE UNDERWATER ENCLOSURE

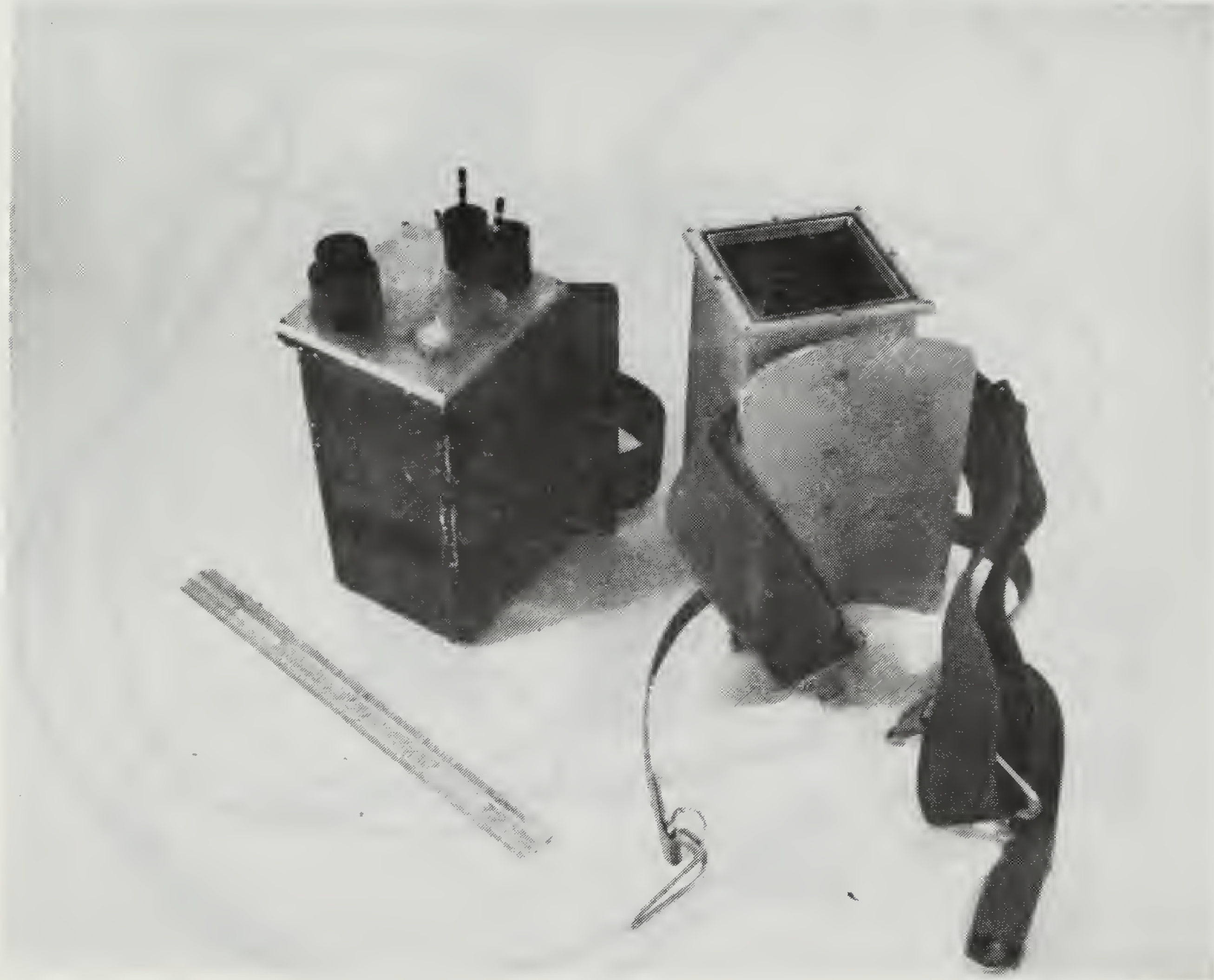


FIGURE 41

THE ENCLOSURE WITH PERIPHERAL EQUIPMENT

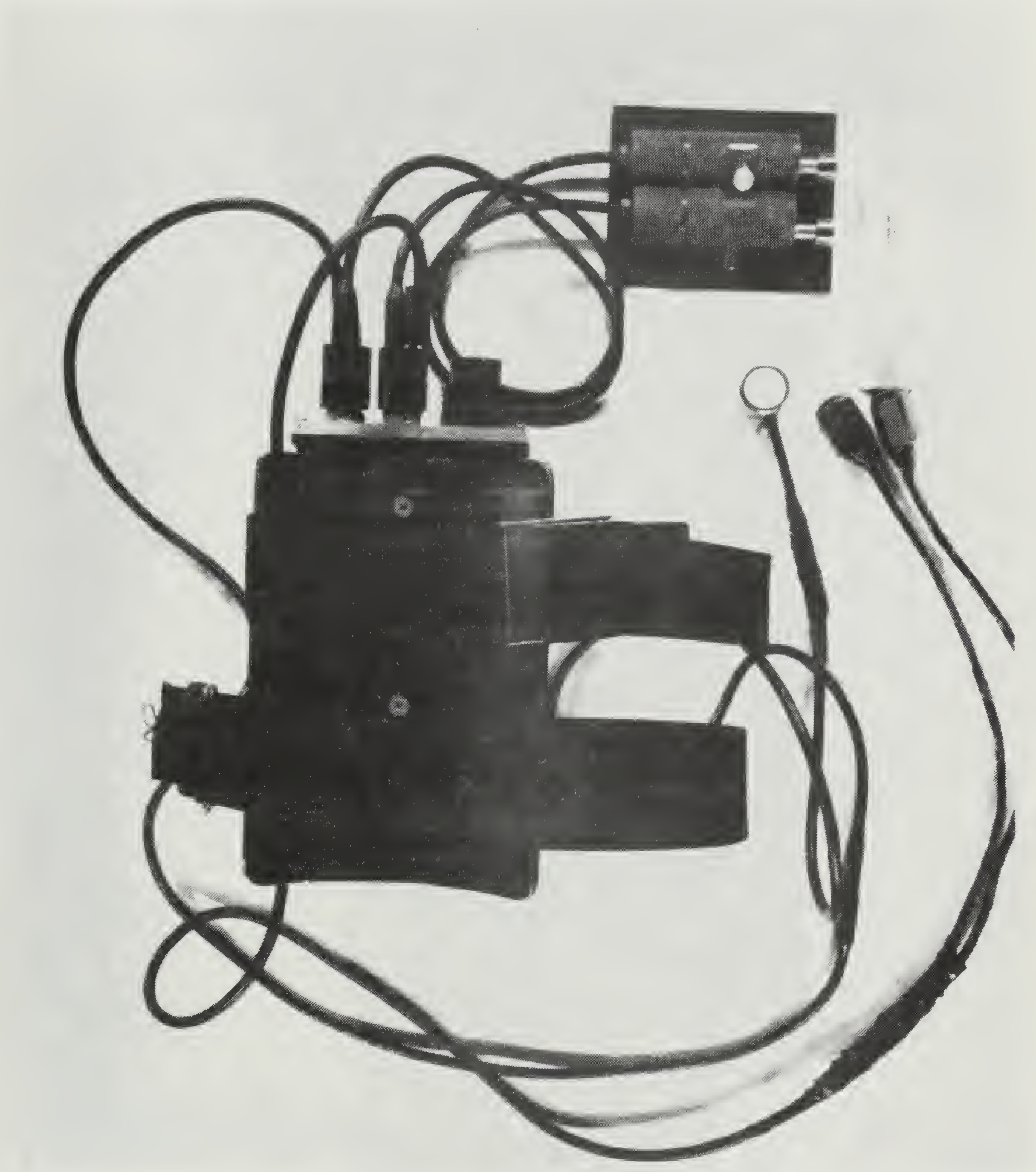


FIGURE 42

CIRCUIT MOUNTING

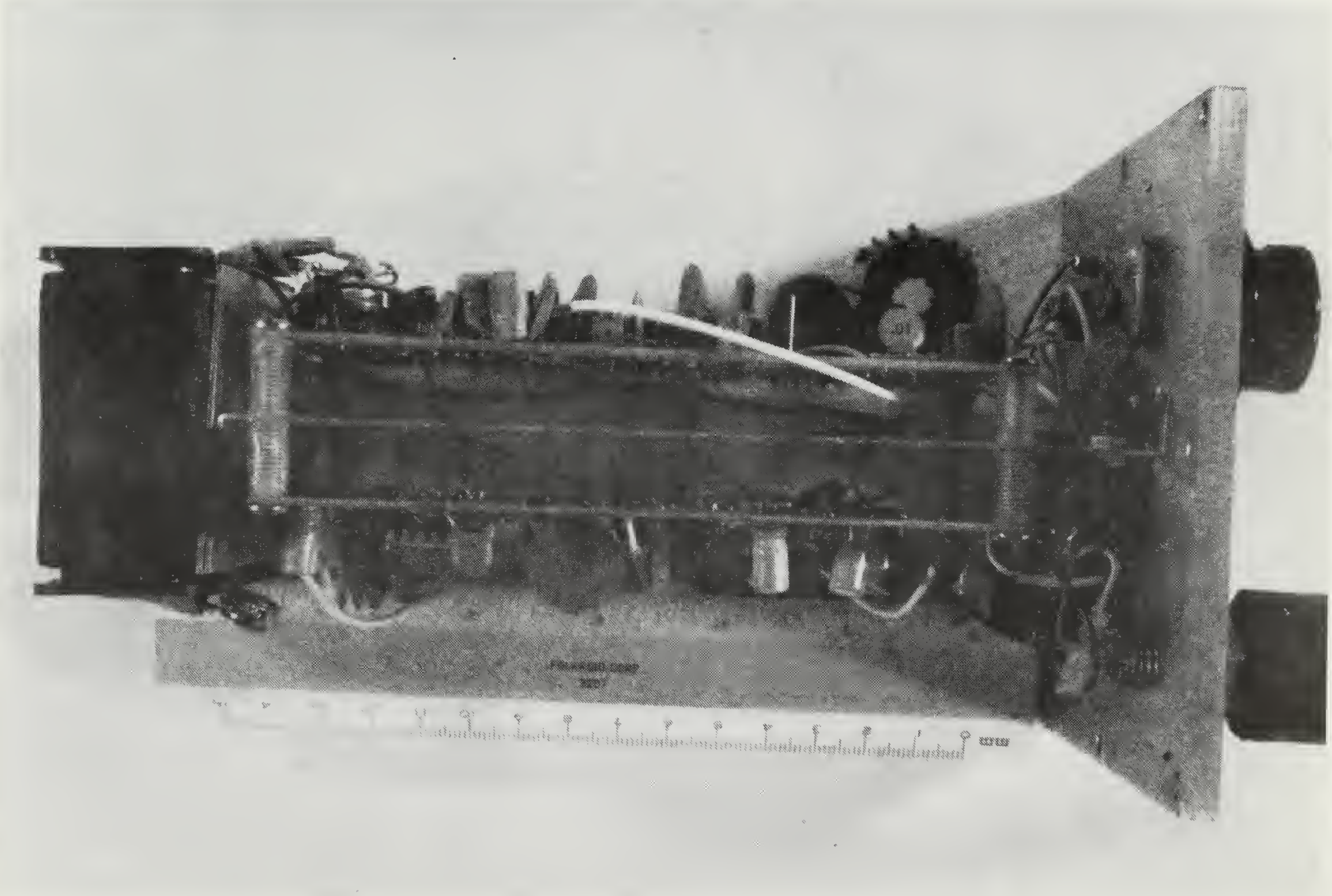


FIGURE 43

TRANSMITTER CIRCUIT BOARD

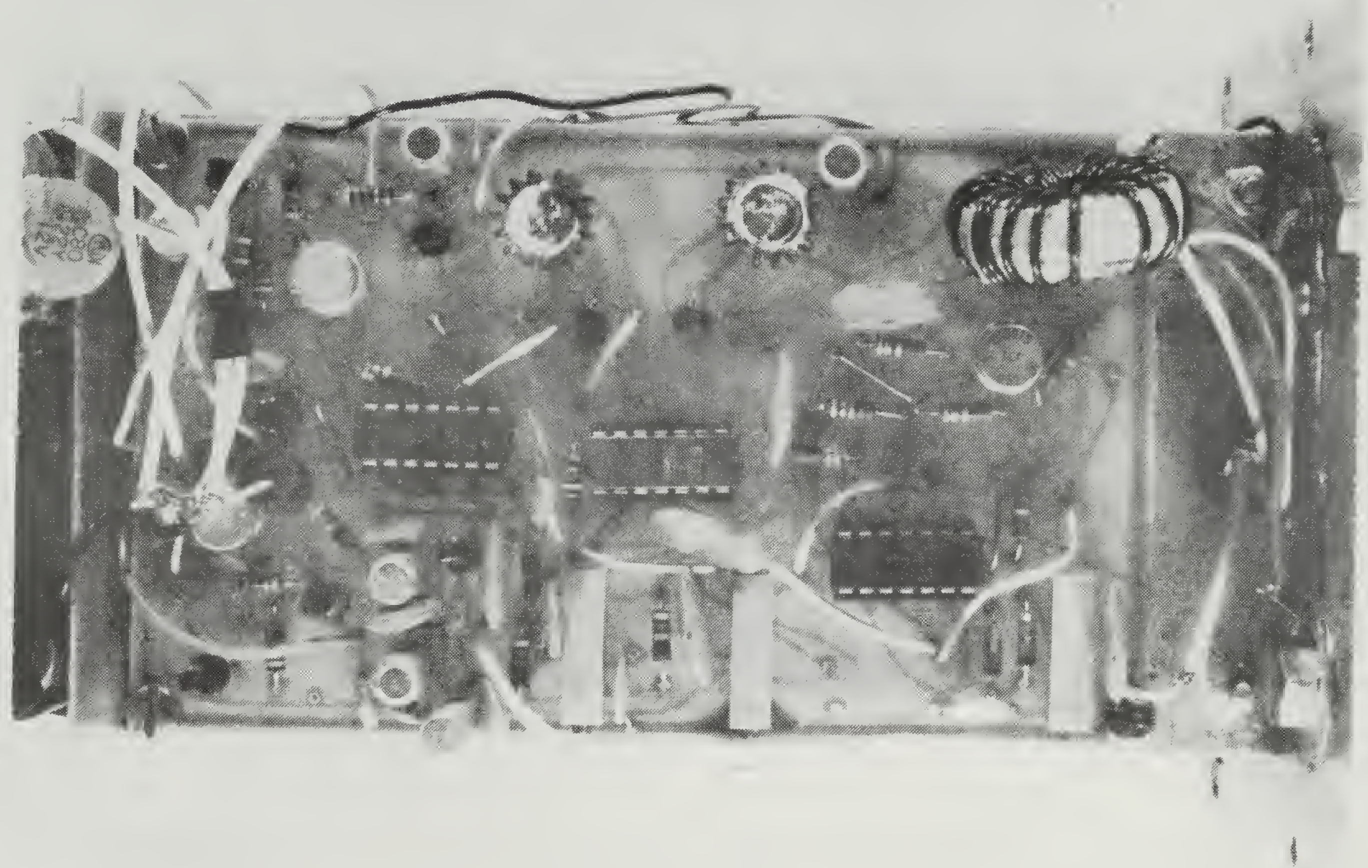


FIGURE 44

RECEIVER CIRCUIT BOARD

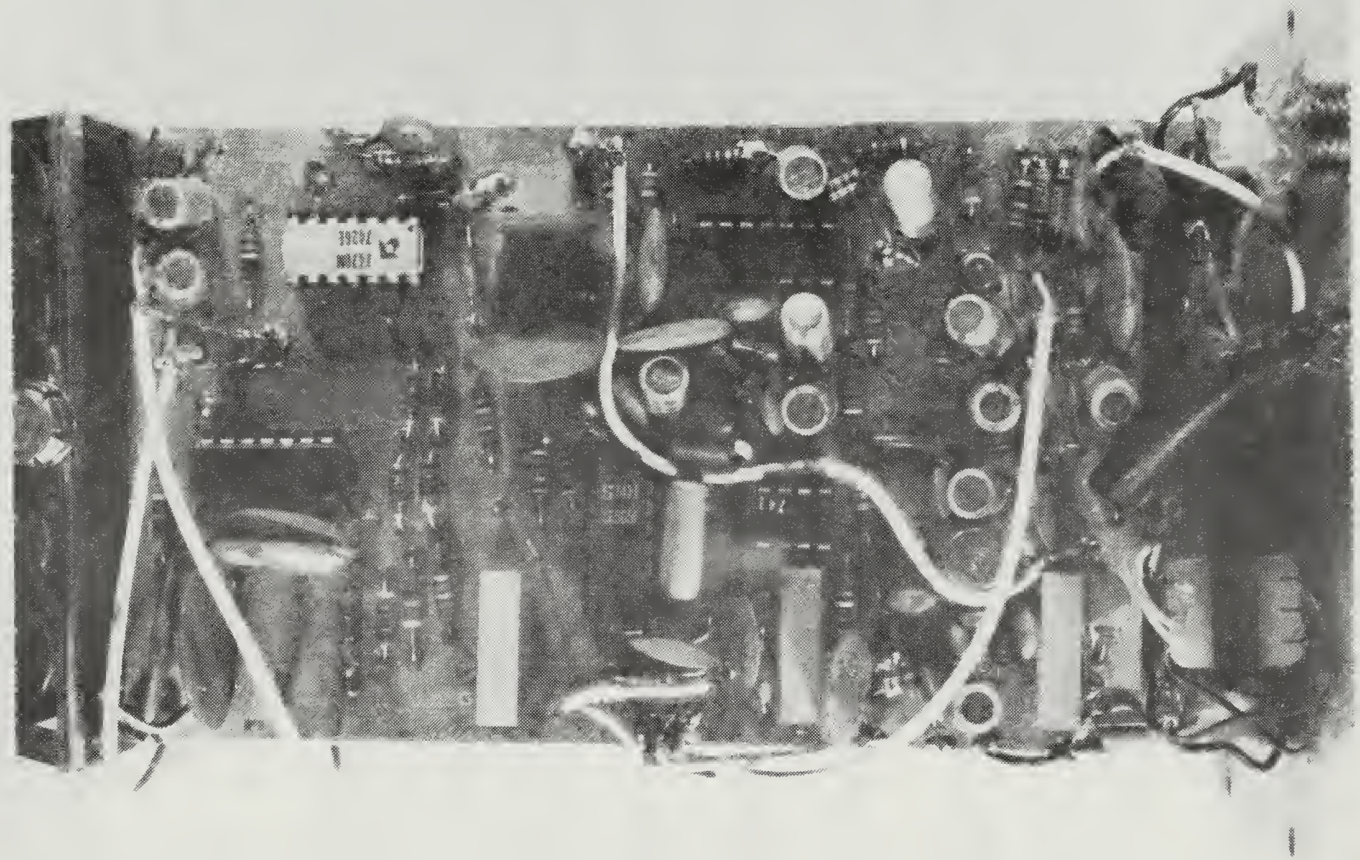


FIGURE 45

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